Adobe® Audition™ 1.5

User Guide

Adobe®

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Adobe® Audition™ 1.5 User Guide for Windows®.

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ug.book Page iii Tuesday, March 16, 2004 1:29 PM

Contents

Learning about Adobe Audition

Getting help 1

Working with Adobe Audition 3

What's New in Adobe Audition 1.5

Use integrated tools 5

Sound your best 6

Work efficiently 7

Chapter 1 **Looking at the Work Area**

About the work area 9

About using Edit View and Multitrack View 10

Switching between views 11

Choosing commands 12

Using toolbars 13

Using windows 14

Navigating in the display window 17

Using the status bar 21

Undoing and redoing changes 22

Organizing files and effects 24

Working with effects 28

Chapter 2 **Setting up Adobe Audition**

About setting up Adobe Audition 35

Setting up devices 36

Setting Adobe Audition preferences 43

Managing temporary files 57

Downloaded from www.Manualslib.com manuals search engine

ug.book Page iv Tuesday, March 16, 2004 1:29 PM

Chapter 3 **Importing, Recording, and Playing Audio**
Opening audio files and multitrack sessions 61
Inserting audio files into multitrack sessions 63
Importing audio from CD 64
Setting the current-time indicator 68
Monitoring time 69
Using the transport controls 71
Recording audio 72
Playing audio 75
Stopping, pausing, and adjusting the playback cursor 78
Monitoring recording and playback levels 79

Chapter 4 **Editing Audio**
About editing audio 83
Creating new audio files 84
Viewing waveforms 85
Selecting audio data 87
Copying, cutting, pasting, and deleting 92
Working with cues 96
Creating play lists 102
Creating and deleting silence 103
Inverting and reversing audio 106
Generating audio 106
Converting the sample type 110
Adding file properties 116

Chapter 5 **Enhancing and Restoring Audio**
About enhancing and restoring audio 117
About the mastering process 117
Analyzing frequency, phase, and dynamic range 118
Removing noise 125
Filtering audio 129
Optimizing amplitude 134

Downloaded from www.Manualslib.com manuals search engine

ug.book Page v Tuesday, March 16, 2004 1:29 PM

Chapter 6 **Applying Stereo, Pitch, and Delay Effects**
About using stereo, pitch, and delay effects 139
Changing stereo imagery 139
Using chorus, flanger, and phaser effects 143
Changing pitch 145
Adding delays and echoes 147
Adding reverb 150
Creating special effects 153
Using multitrack-only effects 156

Chapter 7 **Mixing Multitrack Sessions**
About mixing multitrack sessions 161
Working with sessions 162
Setting advanced session properties 167
Working with clips 168
Working with audio tracks 179
Working with ReWire tracks 184
Working with MIDI tracks 185
Using real-time effects 185
Automating mixes with clip envelopes 188
Using the Mixers window 190
Mixing down ReWire tracks and specific audio clips 196

Chapter 8 **Using Loops**
About loops 197
Defining loops 198
Calculating the tempo of selected ranges 199
Setting permanent loop properties in Edit View 200
Setting impermanent loop properties in Multitrack View 202
Setting the tempo, time signature, and key for sessions 204
Working with loops in the track display 205

Downloaded from www.Manualslib.com manuals search engine

ug.book Page vi Tuesday, March 16, 2004 1:29 PM

Chapter 9 **Working with Video**
About working with video 207
Working with Adobe Premiere Pro and After Effects 207
Importing audio and video from video files 208
Working with video clips 209
Previewing video 210
Preparing video mixdowns for export 211

Chapter 10 **Creating Surround Sound**
About surround sound 213
Using the Multichannel Encoder 214
Panning tracks and buses for surround sound 215
Adjusting volume levels 220
Zooming into and out of the waveform display 221
Previewing the multichannel project 221
Exporting surround-sound files 223

Chapter 11 **Saving, Exporting, and Closing Files**
Saving audio files 227
Saving and exporting sessions 228
Closing files 231
Choosing an audio file format 231

Chapter 12 **Scripting and Batch Processing**
About scripting and batch processing 243
Batch processing cue ranges 243
Normalizing groups of files 244
Batch processing files 247
Working with scripts 249
Using favorites (Edit View only) 253

Downloaded from www.Manualslib.com manuals search engine

ug.book Page vii Tuesday, March 16, 2004 1:29 PM

Chapter 13 **Burning Audio CDs**
Using CD Project View 257
Assembling tracks 257
Editing the source audio for tracks 260
Setting track properties 260
Writing a CD 261

Appendix A **Keyboard Shortcuts**
About keyboard shortcuts 263
Keys for playing audio 263
Keys for selecting ranges, channels, and tracks 264
Keys for copying waveforms 264
Keys for editing clips 265
Keys for repeating commands 265
Keys for using markers 265
Keys for scrolling waveforms and sessions 266
Keys for viewing windows 266

Appendix B **Digital Audio Primer**
Sound fundamentals 267
Waveforms 268
Analog audio 270
Digital audio 270
Sampling rate 270
Bit depth 271
Where Adobe Audition fits into the process 272
Introducing MIDI 273
Conclusion 273

Appendix C **Glossary** 275

Index 291

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ug.book Page 1 Tuesday, March 16, 2004 1:29 PM

Learning about Adobe Audition

Welcome to Adobe® Audition™ 1.5, the ultimate software tool for audio editing, mixing, and mastering.

Adobe provides a variety of options you can use to learn Adobe Audition, including online Help and tool tips. You can also use the Adobe Web site to easily access a wide range of continually updated Web resources, from tutorials to technical support information.

Many files on the Adobe Web site are in Adobe PDF format. To view these files, use Adobe Reader®, included on the Adobe Audition CD.

Getting help

There are a number of ways to get the help you need in Adobe Audition. The following three tables can help you find specific resources related to Adobe Audition features, training resources, and support.

Finding Help for Adobe Audition features	
If you . . .	**Try this . . .**
Want information about installing Adobe Audition	• Insert the Adobe Audition application CD into your CD drive, and follow the on-screen installation instructions. (You cannot run Adobe Audition from the CD.) • See the ReadMe file on the application CD.
Are new to Adobe Audition and want an overview of tools and features	• For information about specific tasks, see "Working with Adobe Audition" on page 3. • For information about the user interface, see "About the work area" on page 9. • Move the pointer over tools and buttons to view tool and button names. • See the beginning tutorials in Help.
Are upgrading from a previous version of Adobe Audition	See "What's New in Adobe Audition 1.5" on page 5 to get an overview of new features. Or, for more detailed information, see the NewFeatures.pdf file on the Adobe Audition application CD.

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ug.book Page 2 Tuesday, March 16, 2004 1:29 PM

Finding Help for Adobe Audition features	
If you . . .	**Try this . . .**
Are looking for detailed information about a feature	• In Help, use the Index or Search tabs. • In windows and dialog boxes, click the Help button or press F1.
Want a list of keyboard shortcuts	See "Keyboard Shortcuts" on page 263.

Finding Adobe Audition training resources	
If you . . .	**Try this . . .**
Want to obtain in-depth Adobe Audition training	• See the tutorials on the Adobe Studio Web site at www.studio.adobe.com. • Browse the Adobe Press materials at www.adobepress.com (English only) and the training resources at www.adobe.com/support/training.html. • For step-by-step lessons, consider the Adobe Classroom in a Book series.
Are looking for background information on digital audio	See the "Glossary" on page 275 and "Digital Audio Primer" on page 267.
Want information about becoming an Adobe Certified Expert	Visit the Partnering with Adobe Web site at http://partners.adobe.com. Certification is available for several different geographical regions.
Want training from an Adobe Certified Training Provider	See the Training page of the Adobe Web site at www.adobe.com/support/training.html.

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ug.book Page 3 Tuesday, March 16, 2004 1:29 PM

Finding support for Adobe Audition	
If you . . .	**Try this . . .**
Want customer or technical support	• Refer to the technical support card provided with your software. • See the Adobe Audition support page at www.adobe.com/support/products/audition.html. • See the ReadMe file installed with Adobe Audition for information that became available after this guide went to press.
Want answers to common troubleshooting questions	Visit the Adobe Audition support page at www.adobe.com/support/products/audition.html.
Want to register your copy of Adobe Audition	• When you first start Adobe Audition, you're prompted to register online. Fill out the form, and then submit it directly or fax a printed copy. • Fill out and return the registration card included with your software package.
Want to access downloads or links to user forums	Visit the main Adobe Audition page at www.adobe.com/audition.

Working with Adobe Audition

You can work with Adobe Audition in many different ways. In this section, you'll find directions to specific information to help you accomplish some common Adobe Audition tasks.

If you want to increase productivity

- Use the Organizer window to quickly organize files, preview loops, and apply effects. (See "Organizing files and effects" on page 24 and "Previewing audio by using the Organizer window" on page 77.)
- Automatically convert audio from a CD into an editable waveform. (See "Importing audio from CD" on page 64.)
- Store selections and start points in cues to speed up editing and navigation tasks. (See "Working with cues" on page 96.)
- Batch process files to quickly apply favorite processing or prepare audio for specific mediums, such as audio CD or the Web. (See "Scripting and Batch Processing" on page 243.)

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ug.book Page 4 Tuesday, March 16, 2004 1:29 PM

If you want to create video soundtracks

- Easily create and remix soundtracks used in Adobe® Premiere® Pro and After Effects® projects. (See "Working with Adobe Premiere Pro and After Effects" on page 207.)
- Time stretch audio clips to match video. (See "Time stretching audio clips" on page 177.)
- Generate noises and tones for sound effects. (See "Generating audio" on page 106.)
- Create surround-sound mixes. (See "About surround sound" on page 213.)

If you want to record and mix musical compositions

- Nondestructively record and edit multitrack sessions of up to 128 tracks. (See "About mixing multitrack sessions" on page 161.)
- Automate mixes with clip envelopes. (See "Automating mixes with clip envelopes" on page 188.)
- Apply, edit, and rearrange real-time effects, without making any permanent changes. (See "Using real-time effects" on page 185.)
- Build compositions with musical loops. (See "About loops" on page 197.)
- Synchronize with ReWire and SMPTE. (See "Setting up ReWire connections" on page 42 and "Setting up for SMPTE synchronization" on page 40.)

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ug.book Page 5 Tuesday, March 16, 2004 1:29 PM

What's New in Adobe Audition 1.5

This overview introduces you to the key new features of Adobe Audition 1.5, including streamlined workflow with other Adobe products, powerful new effects, integrated CD burning, and more.

Use integrated tools

Adobe Audition tightly integrates with flexible audio technology like ReWire and VST, and video applications like Adobe Premiere Pro and Adobe After Effects.

ReWire support Stream full-resolution audio data in real-time between Adobe Audition and other audio software such as Propellerhead Reason and Ableton Live. (See "Setting up ReWire connections" on page 42.)

VST plug-in support Expand your options with integrated support for third-party VST plug-ins, which can also be used in Adobe Premiere Pro. (See "Using plug-in effects" on page 32.)

Enhanced video integration Edit video soundtracks with ease. View video frames in the track display, and import a wide range of video file formats, including AVI, MPEG, and WMV. (See "About working with video" on page 207.)

Improved workflow with other Adobe products Work smoothly with Adobe Premiere Pro, Adobe After Effects, and Adobe® Encore™ DVD by using similar tools, menus, and keyboard shortcuts. (See "Working with Adobe Premiere Pro and After Effects" on page 207.)

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ug.book Page 6 Tuesday, March 16, 2004 1:29 PM

Sound your best

With high fidelity, 32-bit internal processing, Adobe Audition supports up to 32-bit files and sample rates up to 10 MHz. Powerful effects, restoration, and pitch correction tools let you create the exact sound you're after.

Pitch correction tool Correct off-pitch performances and create pitch-based effects. Use automatic mode for quick results, or manual mode for precise control. (See "Using the Pitch Correction effect (Edit View only)" on page 146.)

Frequency space editing Visually isolate, select, and modify sounds in frequency and time using the Marquee Selection tool. (See "Selecting audio frequencies in Spectral View" on page 88.)

Automatic elimination of clicks and pops Quickly and easily restore digital recordings of vinyl source material, wireless mics, DV cameras, and other production audio. (See "Using the Auto Click/Pop Eliminator effect (Edit View only)" on page 125.)

Studio reverb Apply high-quality reverb that conserves processing resources, while offering extensive controls. (See "Using the Studio Reverb effect" on page 153.)

New sample sessions Get up to speed quickly by using any of the 20 sample sessions included with Adobe Audition. Modify the samples to create your own music. (See "About mixing multitrack sessions" on page 161.)

New royalty-free loops Use more than 500 new music loops—for a total of more than 5,000—in a variety of styles including 70's disco, classic rhumba, and wedding and event. (See "About loops" on page 197.)

Vocal extraction Quickly and easily extract the vocal portions of a track to create either a cappella or karaoke-ready tracks, while preserving the stereo image. (See "Using the Center Channel Extractor effect" on page 141.)

Flexible envelope scaling Rescale control points on pan, volume, and effects envelopes to quickly modify a clip in a multitrack mix. Scale all points simultaneously while maintaining relative or absolute relationships between points. (See "Automating mixes with clip envelopes" on page 188.)

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ug.book Page 7 Tuesday, March 16, 2004 1:29 PM

Work efficiently

Adobe Audition puts all the tools you need at your fingertips so you can get your work done quickly and efficiently. An intuitive interface gets you up and running in no time, and integrated editing, mixing, and CD burning streamline your audio workflow.

Integrated CD burning Create masters of your audio compositions by burning gapless audio CDs directly from Adobe Audition. (See "Using CD Project View" on page 257.)

Time stretching Visually drag the edge of any audio clip in a multitrack mix to fit a specific length of time, with or without affecting the clip's pitch. Quickly fit sound effects and dialog to video clips. (See "Time stretching audio clips" on page 177.)

Preroll and postroll playback Speed the process of performing destructive edits and applying effects by listening to the audio preceding and following a selection. (See "Playing audio by using the transport controls" on page 75.)

Custom keyboard shortcut sets Customize keyboard shortcut sets to configure Adobe Audition for your working style. (See "Using shortcuts" on page 12.)

In-time loop previews Use the Organizer window to preview loops in the tempo and pitch of the current session before adding them to your mix. (See "Previewing audio by using the Organizer window" on page 77.)

Task-based documentation Quickly learn how to complete audio production tasks using an updated Help system and user guide organized by subjects such as editing, looping, and video.

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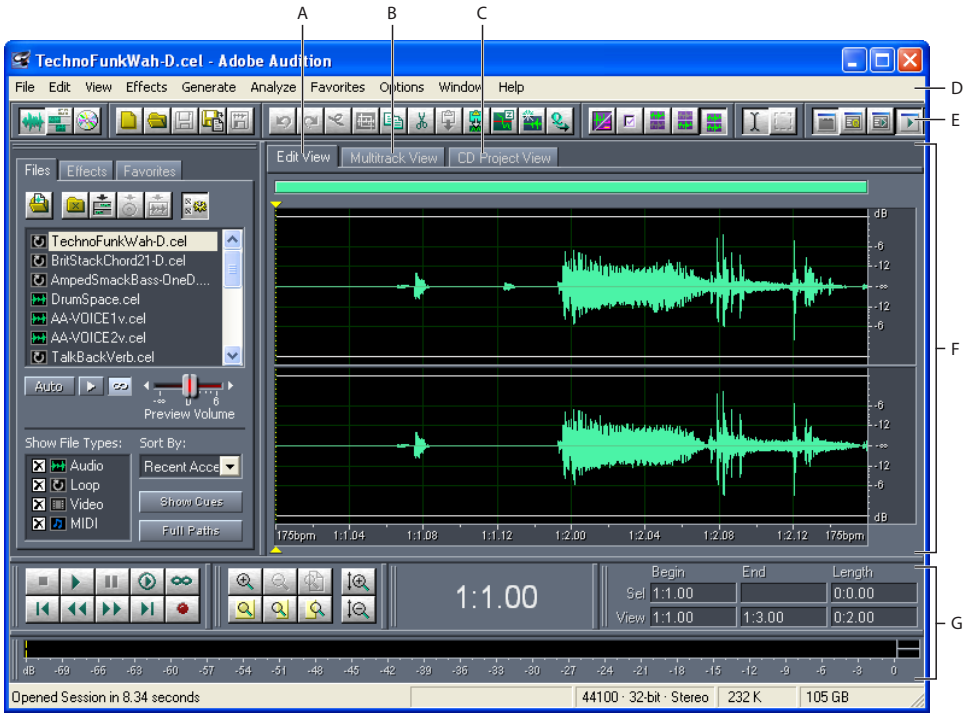
Chapter 1: Looking at the Work Area

Welcome to Adobe Audition. Adobe Audition gives you an efficient work area and user interface to edit and mix audio files.

About the work area

Adobe Audition is divided into three main work areas: Edit View, Multitrack View, and CD Project View. This division is intended to help you focus on the major tasks of editing audio files, mixing sessions, and burning CDs. For more information on the differences between Edit View and Multitrack View, see "About using Edit View and Multitrack View" on page 10. For more information on CD Project View, see "Using CD Project View" on page 257.



Adobe Audition work area
***A.** Edit View tab **B.** Multitrack View tab **C.** CD Project View tab **D.** menus **E.** toolbars **F.** display window **G.** various windows*

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ug.book Page 10 Tuesday, March 16, 2004 1:29 PM

All three views have a similar user interface, including the following components:

Menus The menus in the menu bar contain commands for performing tasks. (See "Choosing commands" on page 12.)

Toolbars The toolbars hold buttons for applying commonly used functions. (See "Using toolbars" on page 13.)

Windows Windows—including the Organizer, Transport Controls, Zoom Controls, Level Meters, and Selection/View Controls—help you monitor and modify audio files. (See "Using windows" on page 14.)

Display window The display window shows you sound in an easy-to-manipulate form. In Edit View, the display window is where you modify single waveforms. In Multitrack View, the display window is where you mix multiple audio files in a session. (See "About editing audio" on page 83 and "About mixing multitrack sessions" on page 161.)

You can change many aspects of Adobe Audition's appearance, including the color scheme, the appearance of buttons, and the appearance of the waveform display, in the Settings dialog box. (See "Setting Adobe Audition preferences" on page 43.)

About using Edit View and Multitrack View

Adobe Audition provides different work areas for editing single waveforms and creating multitrack mixes. To edit single waveforms, you use Edit View. To mix multiple waveforms with MIDI and video files, you use Multitrack View.

Edit View and Multitrack View use different editing methods, and each has unique advantages. Edit View uses a *destructive* method, which changes audio data, permanently altering saved files. Such permanent changes are preferable when converting sample rate and bit depth, mastering, or batch processing. Multitrack View uses a *nondestructive* method, which is impermanent and instantaneous, requiring more processing power, but increasing flexibility. This flexibility is preferable when gradually building and reevaluating a multilayered musical composition or video soundtrack.

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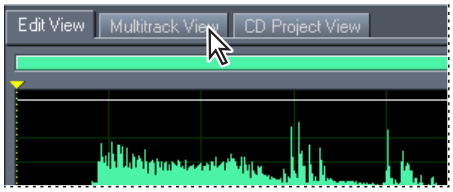
ug.book Page 11 Tuesday, March 16, 2004 1:29 PM

You can combine destructive and nondestructive editing to suit the needs of a project. If a multitrack clip requires destructive editing, for example, simply double-click it to access Edit View. Likewise, if an edited waveform contains recent changes that you dislike, use the Undo command to revert to previous states—destructive edits aren't applied until you save a file. For more information on using Edit View, see "About editing audio" on page 83; for more information on using Multitrack View, see "About mixing multitrack sessions" on page 161.

Switching between views

You can use the tabs above the display window or menu commands to switch between Edit View, Multitrack View, and CD Project View. If you prefer not to use the tabs above the display window, you can hide them.



View tabs above the display window

To switch between views:

Do one of the following:

- Choose View > Edit Waveform View, View > Multitrack View, or View > CD Project View.
- Click the Edit View tab, the Multitrack View tab, or the CD Project View tab above the display window.
- Click the Edit Waveform View button, Multitrack View button, or CD Project View button in the View toolbar. (See "Using toolbars" on page 13.)
- In Multitrack View, double-click a file in the Files tab of the Organizer window or select a file and click the Edit File button. Alternatively, double-click a waveform block in the display window.

To show and hide view tabs above the display window:

Choose View > Show View Tabs. A check mark indicates that the tabs are showing.

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Choosing commands

Commands let you perform a wide variety of tasks. You can choose commands from the menus at the top of your screen or click buttons in a toolbar. You can also use context-sensitive (right-click) menus and keyboard shortcuts to quickly execute commands.

Using context-sensitive menus

Adobe Audition makes liberal use of context-sensitive menus. Whenever you see a simple function button, control, window, or waveform action, try right-clicking it. Chances are you'll be surprised by a useful shortcut menu or a set of handy options that can make Adobe Audition's operation even easier.

Using shortcuts

Adobe Audition provides a set of standard keyboard shortcuts to help you speed up the editing process. For example, instead of using your mouse to go to the Edit menu and choose the Cut command, you can simply press Ctrl + X to cut the selected portion of a waveform. When available, the keyboard shortcut appears to the right of the command name in the menu or in the tool tip for a button or icon. Adobe Audition also provides keyboard shortcuts for performing certain mouse actions. These shortcuts are listed in the Keyboard Shortcuts appendix.

If a shortcut isn't working, it's likely that the window you're trying to run the shortcut in doesn't have focus. For example, if you're in Edit View and you push F11 to bring up the Convert Sample Type dialog box and nothing happens, the waveform display probably isn't the active window. Click the waveform display to give it focus, and then try the shortcut again.

You can change nearly all of the default shortcuts and add shortcuts for other functions. In addition, you can add shortcuts that let you execute commands using keys on a MIDI keyboard, a sequencer, or any other device capable of issuing a MIDI command. This type of shortcut is referred to as a MIDI Trigger. For example, you can assign the Play command in Adobe Audition to the C4 note on your MIDI keyboard.

To enable MIDI triggering:

Choose Options > MIDI Trigger Enable. A check mark indicates the MIDI triggering is on.

Important: *Before attempting to enable MIDI triggering, you must choose a device for MIDI In that's recognized by Windows. For more information, see "Designating which devices you want to use" on page 36.*

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To customize a shortcut:

1 Choose Options > Keyboard Shortcuts And MIDI Trigger.

2 Select the function you want to assign the shortcut to.

Note: *You can filter the list of functions by choosing an option from the Category menu and clicking the Multitrack View or Edit View button. To show all functions, choose (show all) from the Category menu, and deselect the Multitrack View and Edit View buttons.*

3 Do any of the following:

- To assign a keyboard shortcut to the function, click in the Keyboard Shortcut text box and press the desired keyboard combination. Many Adobe Audition users find single key shortcuts (such as *n* for Normalize) faster to use and easier to remember.
- To assign a MIDI trigger to the function, click in the MIDI Trigger text box and press the desired key on the MIDI keyboard. You can also apply MIDI events other than pressing keys (such as pressing the foot pedal).
- To remove a keyboard shortcut or MIDI trigger from the function, click Clear.

4 If you enter a key combination that's already in use, Adobe Audition notifies you of the conflict in the Conflicting Keys text box. Click Clear, and enter a different shortcut before continuing.

5 Click OK.

To restore the default keyboard shortcuts:

1 Choose Options > Keyboard Shortcuts And MIDI Trigger.

2 Choose Adobe Audition Default from the Set list, and click OK.

Using toolbars

Many of Adobe Audition's most commonly used functions are represented as buttons within toolbars, which appear near the top of the main interface. These buttons give you instant access to effects, file handling functions, viewing options, and more, at the press of a button.

To see what a button does, hold your mouse pointer over it to display a tool tip that describes the function in simple terms.

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To show or hide a toolbar:

Choose View > Toolbars, and choose a toolbar name from the submenu. A check mark indicates that the toolbar is showing.

To specify how many rows of buttons are displayed:

Choose View > Toolbars, and choose a number of rows from the submenu.

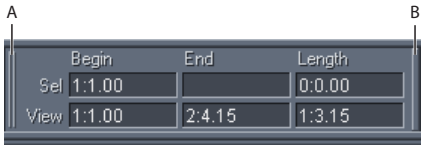
Using windows

Many windows in the Adobe Audition interface can be repositioned and resized to better suit your requirements. You can also hide windows that you're not currently using, and then show them again when needed. For more information on specific windows, see the index or search Help.

Repositioning and resizing windows

When you reposition a window, you can dock it in a specific location in the interface, or you can undock the window so that it floats above the main window. To identify docked windows, look for two thin vertical or horizontal lines. These lines are the *handle* (or grab bar) of a docked window. Move your mouse over a handle, and the cursor looks like a plus sign with arrows at each end ✣.

Some docked windows can also be resized. If resizing is possible, the docked window will have a single, thicker horizontal or vertical bar, called a *resize bar*. When you move your mouse over a resize bar, the cursor takes on the appearance of two lines with two arrows ⇹.



Docked window
***A.** Handle **B.** Resize bar*

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To undock a window:

Drag the window's handle to the middle of the work area until you see an outline of the window.

The window is now a standard floating window. You can move the window by dragging its title bar.

Press Ctrl while moving a floating window around to force it to not dock. That way you can float the window over an area that it would normally try to dock to. To disable this feature, select Ctrl Key Allows Dockable Windows to Dock in the General tab of the Settings dialog box. (See "Setting Adobe Audition preferences" on page 43.)

To dock a window in a different location:

1 Drag the window's handle around the work area to locate potential docking areas. The resize bars of other docked windows will light up wherever docking is possible.

2 When you locate the desired docking area, release the mouse button. The window snaps into its new location.

If a window is docked in the same row with other windows, you can force the window into a new row by right-clicking the window's handle and selecting Force New Row. Likewise, deselecting Force New Row causes the window to dock in the previous row (if there's room).

To resize a docked window:

Drag the window's resize bar.

Even if the resize bar is visible, resizing might not be possible due to the other windows that are in the row with the window you're trying to resize.

To reset windows to the default layout, select Restore Default Workspace in the General tab of the Settings dialog box. (See "Setting Adobe Audition preferences" on page 43.)

Showing and hiding windows

You can free up space in the work area by closing windows when you aren't using them, and then redisplay the windows as needed. The Window menu lists all available windows; a check mark indicates that a window is currently showing.

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ug.book Page 16 Tuesday, March 16, 2004 1:29 PM

To hide a window:

Do one of the following:

- Choose the window name from the Window menu.
- Click the button that corresponds to the window name in the View toolbar. (See "Using toolbars" on page 13.)
- For docked windows, right-click the window's handle and choose Close.
- For undocked windows, click the *X* button on the window's title bar.

To show a window:

Choose the window name from the Window menu, or click the window's button in the View toolbar.

Using placekeeper windows

Placekeeper windows let you define the aspect ratio of a docking area. For example, if you try docking the Track EQ controls above the transport controls, they end up going underneath the whole session display, which creates a view that isn't very useful (or aesthetically pleasing). You can use a placekeeper, though, on either side of the Track EQ to force the EQ into a certain aspect ratio. You can also use placekeepers just for appearance's sake, just because you like the way they let you customize the work area.

You can create up to four placekeeper windows, and insert them wherever docking is allowed. You can also change the appearance of placekeeper windows by filling them with a pattern.

To insert a placekeeper window:

1 Choose Window > Placekeeper.

2 Dock the placekeeper in the desired location. The window is automatically resized to fit the docked area.

To change the appearance of a placekeeper window:

Right-click the window's handle, and choose a fill option: Nothing, Cool Texture, or Squares. To make future placekeeper windows adopt the current appearance, choose Make Default.

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ug.book Page 17 Tuesday, March 16, 2004 1:29 PM

To delete a placekeeper window:

Right-click the window's handle, and choose Close.

Navigating in the display window

The display window shows you the current waveform (in Edit View) or session (in Multitrack View). You can control how much of the waveform or session is displayed by zooming and scrolling. You can also use the selection and view controls to determine the beginning time, ending time, and length of audio data in the display window.

Zooming

Zooming lets you adjust the view in the display window to best meet your needs. For example, you can zoom in to clearly see the samples in a waveform, or you can zoom out to get a visual overview of a waveform or session.

The Zoom Controls window provides a variety of tools for zooming. You can also zoom by dragging in the horizontal scroll bar, vertical scroll bar (Multitrack View only), or vertical ruler.

Zoom controls

To show or hide the zoom controls:

Do one of the following:

- Choose Window > Zoom Controls. A check mark indicates that the controls are visible.
- Click the Hide/Show Zoom Controls button in the View toolbar. (See "Using toolbars" on page 13.)

If you don't like the default location of the zoom controls, you can reposition them or detach them so they float above the main window. (See "Using windows" on page 14.)

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ug.book Page 18 Tuesday, March 16, 2004 1:29 PM

To zoom in or out by using the zoom controls:

Do any of the following:

- Click the Zoom In Horizontally button to zoom in on the center of the visible waveform window or session.
- Click the Zoom In Vertically button to increase the vertical scale resolution of a waveform's amplitude display (in Edit View) or decrease the number of viewed tracks in the session display (in Multitrack View).
- Click the Zoom To Selection button to zoom in on the actively selected waveform or session range.
- Click the Zoom In To Right Edge Of Selection button to zoom in on the right boundary of the actively selected waveform range or session.
- Click the Zoom In To Left Edge Of Selection button to zoom in on the left boundary of the actively selected waveform range or session.
- Click the Zoom Out Horizontally button to zoom out from the center of the visible waveform window or session.
- Click the Zoom Out Full Both Axis button to zoom out to display the entire waveform or blocks that are contained within a session.
- Click the Zoom Out Vertically button to decrease the vertical scale resolution of a waveform's amplitude display (in Edit View) or to show more tracks in the session display (in Multitrack View).

To zoom in or out by using a scroll bar or ruler:

Do either of the following:

- To change the viewable range of time, position the pointer in the timeline or over the left or right edge of the horizontal scroll bar. Then drag to the left or right. A magnifying glass with arrows icon appears as you drag.
- To change the viewable range of amplitude (in Edit View) or tracks (in Multitrack View), hold down the right mouse button in the vertical ruler, and drag up or down. The magnifying glass with arrows icon appears as you drag.

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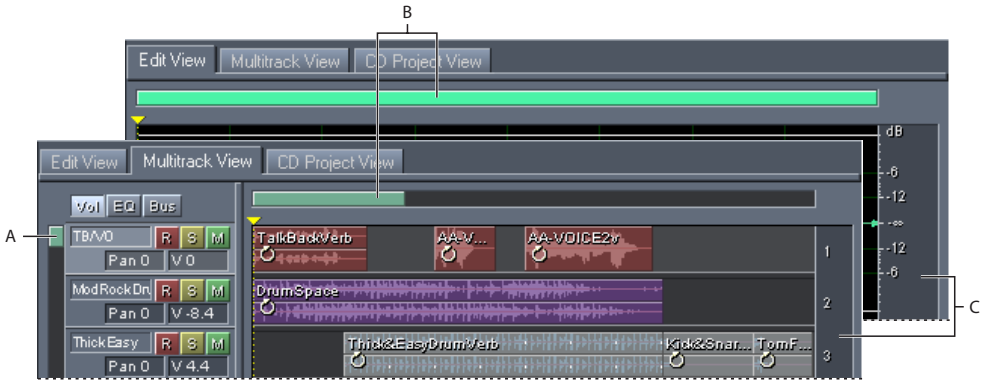
ug.book Page 19 Tuesday, March 16, 2004 1:29 PM

You can also use the wheel on your mouse to zoom in and out. To do so, place the pointer over the horizontal scroll bar, timeline, vertical scroll bar (Multitrack View only), or vertical ruler, and roll the mouse wheel. To set a zoom percentage for the mouse wheel, enter a value for Zoom Factor in the General tab of the Settings dialog box. (See "Setting Adobe Audition preferences" on page 43.)

Scrolling

The display window provides several scrolling devices. The horizontal scroll bar—which, by default, is at the top of the display window—lets you scroll forwards and backwards in time throughout a waveform (in Edit View) or session (in Multitrack View). The vertical ruler on the right side of the display window lets you scroll through amplitude ranges (in Edit View) or tracks (in Multitrack View). In Multitrack View, there's an additional vertical scroll bar on the left side of the display window that lets you scroll through tracks.



Scrolling devices
***A.** Vertical scroll bar **B.** Horizontal scroll bar **C.** Vertical ruler*

To scroll in the display window:

Do either of the following:

- To scroll to the left or right, drag the horizontal scroll bar. Or, click to the left or right of the scroll bar to page through the display one screen at a time.
- To scroll up or down, drag in the vertical ruler. In Multitrack View, you can also drag the vertical scroll bar or click above or below the scroll bar to page through the display one screen at a time.

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ug.book Page 20 Tuesday, March 16, 2004 1:29 PM

You can also use the wheel on your mouse to scroll in the display window. To do so, place the pointer over the display window, and roll the mouse wheel.

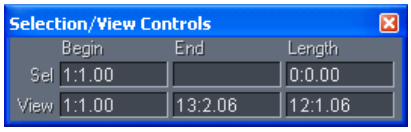
To change the position of the horizontal scroll bar:

Right-click the horizontal scroll bar, and choose a display option: Above Display or Below Display.

Using the selection and view controls

The Selection/View Controls window shows the beginning and ending points, as well as the total length of both the selection and the section of the waveform or session that's currently visible. Both the selection and display range is shown in the current time-display format. For information on changing the time-display format, see "Monitoring time" on page 69.

In addition to viewing time information, you can also use the selection and view controls to adjust selections and change the section of audio data that is visible in the display window. Simply enter new values for Begin, End, and Length. After you click in a text box, you can right-click to access additional context-menu commands.



Selection and view controls

To display the selection and view controls:

Do one of the following:

- Choose Window > Selection/View Controls. A check mark indicates that the window is showing.
- Click the Hide/Show Selection/View Controls button in the View toolbar. (See "Using toolbars" on page 13.)

If you don't like the default location of the selection and view controls, you can reposition them or detach them so they float above the main window. (See "Using windows" on page 14.)

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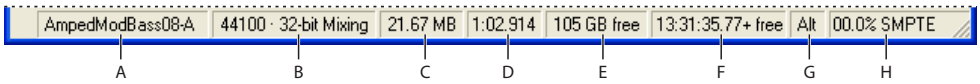
ug.book Page 21 Tuesday, March 16, 2004 1:29 PM

Using the status bar

The status bar runs along the very bottom of Adobe Audition's main window. It can display information such as sample format, file size, and free disk space.



Status bar
***A.** Data Under Cursor **B.** Sample Format **C.** File Size **D.** File Size (time) **E.** Free Space **F.** Free Space (time) **G.** Keyboard Modifiers **H.** SMPTE Slave Stability*

To show or hide the status bar:

Do one of the following:

- Choose View > Status Bar > Show. A check mark indicates that the status bar is visible.
- Click the Hide/Show Status Bar button in the View toolbar. (See "Using toolbars" on page 13.)

To change the type of information that is displayed in the Status Bar:

Choose View > Status Bar or right-click the Status Bar, and select the desired display options. Selected items appear in the Status Bar; unselected items are hidden.

You can choose from the following options:

Data Under Cursor Shows useful information such as the channel (if a current waveform is stereo), the amplitude (measured in decibels), and the time (hours:minutes:seconds:hundredths of seconds) from the beginning of the audio file. This data is computed at the precise point where your mouse pointer is placed within the wave display, and changes dynamically when you move the pointer. For example, if you see *R: –15.2 dB @ 0:00:242* in the Status Bar when in Edit View, this means that your pointer is over the right channel at 0.242 seconds into the waveform, and the amplitude at that precise point is –15.2 dB.

In the Multitrack View, you'll see even more beneficial data such as Pan and Volume envelope positions, envelope positions for effects envelopes, dynamic effect settings, and the current position of the wave block as you drag it around.

Sample Format Displays sample information about the currently opened waveform. For example, a 44,100 kHz 16-bit stereo file shows up as *44100 – 16-bit – stereo*.

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ug.book Page 22 Tuesday, March 16, 2004 1:29 PM

File Size) Represents how large the active audio file is, measured in kilobytes. If you see *308 K* in the Status Bar, then the current waveform or session is 308 kilobytes (KB) in size.

File Size (time) Shows you the length (measured in time) of the current waveform or session. For example, *0:01:247* means the waveform or session is 1.247 seconds long.

Free Space In Edit View and Multitrack View, shows how much space is available on your hard drive. In CD Project View, shows how much space remains on a CD based on which View menu item is selected: 74 min CD or 80 min CD.

Free Space (time) In Edit View and Multitrack View, displays the amount of available time left for recording, based upon the currently selected sample rate. This value is shown as minutes, seconds, and thousandths of seconds. For example, if Adobe Audition is set to record an 8-bit mono waveform at 11,025 kHz, the time left might read something like *4399:15.527 free*. Change the recording options to 16-bit stereo at 44,100 kHz, and the remaining time value becomes *680:44.736 free*.

In CD Project View, shows how much space remains on a CD based on which View menu item is selected: 74 min CD or 80 min CD.

Keyboard Modifiers Displays the status of your keyboard's Ctrl, Shift, and Alt keys.

SMPTE Slave Stability Indicates the stability of incoming SMPTE timecode compared to Adobe Audition's internal clock. For example, *95.0% SMPTE* indicate a very strong SMPTE signal. Percentages above 80% should be stable enough to maintain sync. For more information on SMPTE synchronization, see "Setting up for SMPTE synchronization" on page 40 and "Using sessions as SMPTE masters or slaves" on page 166.

Undoing and redoing changes

Adobe Audition keeps track of the edits you perform during the course of an editing session. These changes are stored in a temporary file on your hard drive. They aren't permanently applied to the file until you save and close it, giving you unlimited undo and redo capability.

When you work with very large audio files, you might not have enough free disk space to save the Undo data before continuing with an edit. In addition, the time required to save the Undo information might slow down your work. You can solve either problem by disabling the Undo function.

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ug.book Page 23 Tuesday, March 16, 2004 1:29 PM

To undo a change:

Choose Edit > Undo *[name of change]*. Or, click the Undo button in the toolbar.

The Undo command conveniently indicates which change you're undoing. For example, it may appear as Undo Delete or Undo Normalize. If you haven't yet edited a waveform, or if Undo is disabled, this command appears as Can't Undo.

If you forgot which editing action you last performed on a waveform, look at the Undo command to refresh your memory, whether you want to undo the action or not.

To discard edits made since you last saved the file:

In Edit View, choose File > Revert To Saved.

To redo a change:

In Edit View, choose Edit > Redo *[name of change]*. Or, click the Redo button in the toolbar.

To repeat the last command:

In Edit View, choose Edit > Repeat Last Command. You can repeat most editing functions in Adobe Audition by using this command; however, there are a few exceptions (such as Delete).

To disable or enable the Undo function:

Do one of the following:

- In Edit View, choose Edit > Enable Undo/Redo. A check mark indicates that the Undo function is enabled.
- Choose Options > Settings, and click the System tab. Select or deselect Enable Undo, and click OK. You can also specify the minimum number of undo levels, and you can purge all undo files. (See "System options" on page 45.)

If you don't have enough disk space to save the undo information, you can change the Temp folder to a different drive, if available.

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ug.book Page 24 Tuesday, March 16, 2004 1:29 PM

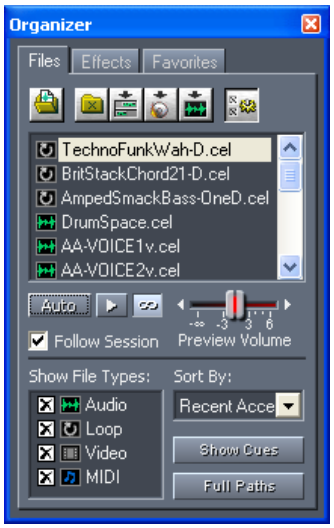
Organizing files and effects

The Organizer window appears in Edit View, Multitrack View, and CD Project View. This handy, tabbed window lets you easily open and close files, see a list of all open waveforms and MIDI files, choose effects with ease, and more. By default, the Organizer window is docked to the left of the waveform or session display; however, you can reposition it or detach it so it floats above the main window. (See "Using windows" on page 14.)

Organizing files

The Files tab in the Organizer window displays a list of open waveforms, MIDI files, and video files. You can use the Files tab to import files, select files for editing, insert clips into sessions, insert tracks into CDs, and close files.

The Files tab also provides a variety of advanced options that let you show and hide cues, change the listing and sort order of files, and play files. You can choose to hide advanced options if you don't use them.



Files tab in the Organizer window

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ug.book Page 25 Tuesday, March 16, 2004 1:29 PM

To display the Files tab:

1 If the Organizer window isn't showing, choose Window > Organizer to display it.

2 Click the Files tab in the Organizer window. The following buttons appear at the top of the Files tab:

- The Import File button lets you import audio, MIDI, and video files into Adobe Audition.
- The Close Files button lets you close all selected files in the Files tab.
- The Insert Into Multitrack button lets you insert all selected files, each into their own track, in Multitrack View. (See "Inserting audio files into multitrack sessions" on page 63.)
- The Insert Into CD Project button lets you insert all selected files into CD Project View. (See "Inserting tracks" on page 258.)
- The Edit File button lets you open the selected file in Edit View. (See "Switching between views" on page 11.)

To select files in the Files tab:

Do any of the following:

- To select a single file, click it.
- To select adjacent (or *contiguous*) files, click the first file in the desired range, and then Shift-click the last.
- To select nonadjacent (or *noncontiguous*) files, Ctrl-click them.

Note: *If you select multiple files, only the last file you click appears in Edit View.*

To show or hide advanced options in the Files tab:

Click the Advanced Options button at the top of the Files tab. When showing, the advanced options appear at the bottom of the Files tab.

For information on the play controls in the Files tab, see "Previewing audio by using the Organizer window" on page 77.

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ug.book Page 26 Tuesday, March 16, 2004 1:29 PM

To change the listing and sort order of files in the Files tab:

Make sure that the advanced options are showing, and do any of the following:

- To show or hide files, select a Show File Types option. An *X* indicates that files of the specified type are showing.
- To change the sort order of files, choose an option from the Sort By menu.
- To display the full path *[drive, folder(s), filename]* of the entries in the File tab, select the Full Path button. To display only the filenames, deselect this button.

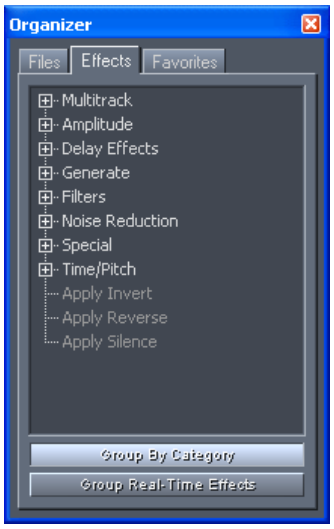
To show or hide cues in the Files tab:

Make sure that the advanced options are showing, and click Show Cues.

When Show Cues is selected, a plus icon appears next to files that contain cues. Click the plus icon to display the cue names. For more information on cues, see "Working with cues" on page 96.

Organizing effects

The Effects tab in the Organizer window lists all of the effects at your disposal. The listing includes all of Adobe Audition's effects as well as all installed DirectX and VST audio plug-ins. You can change the grouping of effects to best meet your needs.



Effects tab in the Organizer window

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ug.book Page 27 Tuesday, March 16, 2004 1:29 PM

To display the Effects tab:

1 If the Organizer window isn't showing, choose Window > Organizer to display it.

2 Click the Effects tab in the Organizer window.

To change how the effects are grouped:

Click the buttons at the bottom of the Effects tab:

- Select Group By Category to list effects in a hierarchy where categories and their entries are shown in the same order as they appear in the Effects menu.
- Deselect Group By Category to display all effects in roughly the same order as they appear in the Effects and Generate menus.
- Select Group Real-Time Effects to list effects in a hierarchy where all of the Real-Time Effects are grouped together, the Off-Line Effects are grouped together, and the Multi-track Effects are grouped together.
- Deselect Group Real-Time Effects to return to the previous view.

Organizing favorites

Favorites are effects, scripts, and even third-party tools that you've saved for easy access. The Favorites tab in the Organizer window lists all of the favorites you've created. (These same items are listed in the Favorites menu.)

Favorites tab in the Organizer window

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To display the Favorites tab:

1 If the Organizer window isn't showing, choose Window > Organizer to display it.

2 Click the Favorites tab in the Organizer window.

For more information on creating and editing favorites, see "Using favorites (Edit View only)" on page 253.

Working with effects

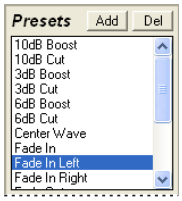
Effects provide much of the functionality in Adobe Audition. For example, you use effects to remove noise, optimize volume, change pitch, and add reverb. If Adobe Audition doesn't provide the effect you want, you may be able to purchase a plug-in effect to do the job.

As you apply effects, you'll notice similarities between Adobe Audition's effect dialog boxes. For example, many effect dialog boxes provide presets for storing and recalling your favorite settings. Some effect dialog boxes also provide graph controls for adjusting settings. As you adjust settings, you can use the Preview option to preview effects in real time.

For information on using specific effects, search for the effect name in Help or look in the index.

Using presets

Many of Adobe Audition's effects and other functions have presets that are available for easily storing and recalling your favorite settings. You can add and remove presets at any time.



Presets in the Amplify/Fade dialog box

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To apply a preset:

Double-click the preset name. The settings defined by the preset are reflected in the dialog box.

To add a preset:

1 Adjust the effect settings as desired.

2 Click Add in the Presets area of the effect dialog box.

3 Enter a name for the preset, and click OK. Your new preset is added to the list of other presets, which is automatically sorted alphabetically.

To modify a preset:

1 Double-click the preset name, and adjust the settings as desired.

2 Click Add, enter the name of the current preset, and click OK.

3 Click OK when prompted to replace the preset.

To remove a preset

Select the preset, and click Del.

Using graph controls

Many of Adobe Audition's effects use graph controls for adjusting parameters. By adding and moving control points on the graph, you can tailor the effect to precisely meet your needs.

By default, graphs display straight lines between control points. However, some graphs provide a Splines or Spline Curves option for generating a curve between control points. Using spline curves lets you create smoother transitions between points.

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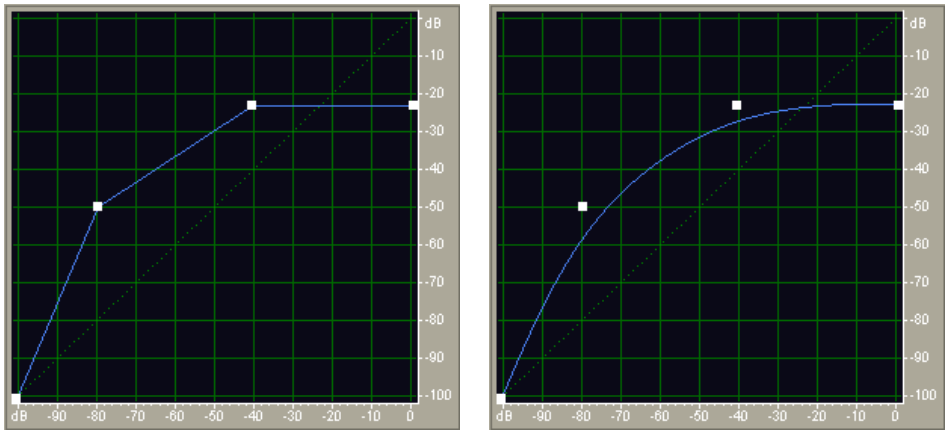
ug.book Page 30 Tuesday, March 16, 2004 1:29 PM



Graph with straight lines between control points compared to graph with spline curves

When you use spline curves, the line won't travel directly through the control points. Instead, the points control the shape of the curve. To get the curve closer to a control point, click to create more control points near the point in question. The more control points there are clustered together, the closer the spline curve will be to those points.

To use graph controls:

Do any of the following:

- To add a control point to the graph, click in the grid at the location where you want to place the point.
- To enter the values for a control point numerically, right-click the point to bring up the edit box, or double-click the graph's curve.
- To move a point on the graph, drag it to a new location.
- To remove a point from the graph, drag it off the graph.

Note: *When the pointer is located over a control point, you'll see it change from an arrow to a hand.*

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Previewing effects in Edit View

Many dialog boxes provide a Preview button for previewing effects in real time. This means that you can monitor the processed signal before applying the effect to the waveform. The preview feature updates in real time, meaning that changes you make to effect settings while in the dialog box for that effect become audible immediately, while the audio is playing.

Keep in mind that your system's performance affects the preview feature. On slower systems, some effects may tend to break up or skip during preview. In Multitrack View, the preview is not necessary, as effects are used nondestructively. Basically, every effect in the Multitrack View is in preview all the time. For more information on the differences between destructive and nondestructive editing, see "About using Edit View and Multitrack View" on page 10.

In Edit View, you can add an optional preroll or postroll amount to the duration of the preview. This is especially useful when previewing effects for small ranges and marquee selections because it lets you hear how the in and out transitions are affected by the effects settings.

To preview effects in real time:

1 Click the Preview button to start playing the audio.

2 Adjust the effect settings as desired.

3 To compare the original audio to the processed audio, select and deselect the Bypass option. When the option is selected, you hear the original audio; when the option is deselected, you hear the processed audio.

4 When you're satisfied with the settings, click Stop.

To add a preroll and postroll duration to a preview:

1 In Edit View, right-click the Play button ▸ or the Play To End button ⏵ in the transport controls, and choose Preroll And Postroll Options.

2 In the Effects Preview section of the Preroll And Postroll Options dialog box, enter durations for the preroll and postroll, and click OK.

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ug.book Page 32 Tuesday, March 16, 2004 1:29 PM

3 Do one of the following:

- Choose Effects > Enable Preroll And Postroll Preview.
- In an effects dialog box, select Enable Preroll And Postroll Preview. This option appears below the Presets. If a dialog box does not have Preset, the Enable Preroll And Postroll Preview option will not appear; however, you can still enable preroll and postroll preview by choosing Effects > Enable Preroll And Postroll Preview.

4 Preview an effect as described in the previous procedure.

Using plug-in effects

DirectX and VST plug-ins let you extend the already powerful effects at your disposal in Adobe Audition. Before you can start using plug-in effects, you must set them up in Adobe Audition. For DirectX effects, this process involves enabling the effects and then refreshing the effects list. For VST effects, you need to verify that Adobe Audition is scanning the directories where the effects are installed; then, you must refresh the effects list.

After that, using plug-in effects is as easy as using any other Adobe Audition effect. Just select an area to process, and choose the effect from the Effects > DirectX or Effects > VST menu (or from the Effects tab of the Organizer Window). Of course, you'll need to consult the documentation provided by the plug-in manufacturer for any help with its features.

Note: *If Adobe Premiere Pro and Adobe Audition are installed on the same computer, Adobe Audition automatically displays the VST plug-ins that come with Adobe Premiere Pro.*

To enable DirectX effects:

Do one of the following:

- In Edit View, choose Effects > Enable DirectX Effects.
- In Multitrack View, click the FX button in the track controls. In the Track Effects Rack dialog box, click Enable DirectX Effects, and then click OK.

This causes Adobe Audition to scan your system for DirectX plug-ins. After the plug-ins are activated, the Enable DirectX Effects option is removed from the menu and dialog box.

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To set up directories for VST effects:

1 In Edit View, choose Effects > Add/Remove VST Directory.

The Add/Remove VST Directory lists the directories that Adobe Audition will scan for VST plug-ins when you choose Effects > Refresh Effects List.

2 Do either of the following:

- To add a new directory, click Add, locate or create the folder you want Adobe Audition to scan for VST plug-ins, and click OK.
- To remove a directory, select the directory and click Remove.

To refresh the effects list after installing new effects:

In Edit View, choose Effects > Refresh Effects List.

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ug.book Page 35 Tuesday, March 16, 2004 1:29 PM

Chapter 2: Setting up Adobe Audition

You can customize the way Adobe Audition works by setting up devices and internal preferences.

About setting up Adobe Audition

Setup tasks fall into several categories. Perhaps the most important is setting up the devices you want to use with Adobe Audition. If you have multiple sound cards, or a single card that has multiple inputs and outputs, you need to specify which devices you want to use for playback and recording. In addition, you can set up MIDI devices, external controllers, and ReWire connections for use with Adobe Audition. For more information on these tasks, see "Setting up devices" on page 36.

Another category of setup tasks is customizing internal Adobe Audition preferences to best suit your needs. For example, you can change the appearance of the workspace, set buffer sizes to optimize performance, change the locations of temporary folders to better utilize disk space, and customize the wave and session displays. For more information on these tasks, see "Setting Adobe Audition preferences" on page 43.

A final category of setup tasks is managing the size of temporary files. The size of temporary files is limited only by the amount of disk space that is available; however, when you're working with very large files (or when you have many files open at the same time), your disk space may run low. If this happens, you can delete temporary files you're not using, clear specific Undo items, and change the amount of reserve space. For more information on these tasks, see "Managing temporary files" on page 57.

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ug.book Page 36 Tuesday, March 16, 2004 1:29 PM

Setting up devices

You can use a wide range of devices with Adobe Audition. Sound card inputs let you bring audio signals into Adobe Audition through sources such as microphones, tape decks, and digital effects units. Sound card outputs let you monitor audio signals through sources such as speakers and headphones. MIDI ports let you connect Adobe Audition to MIDI keyboards and synthesizers. You can also synchronize Adobe Audition with ReWire applications and hardware or software components that support SMPTE/MTC timecode.

Designating which devices you want to use

The Device Order dialog box lets you designate which devices you want to use with Adobe Audition. When working in Edit View, you can designate one stereo output device to use for playback and one stereo input device to use for recording. When working in Multitrack View, you can assign different input and output devices to each audio track. However, before you can do this, you must specify which devices you plan to use and the order in which you want to view them.

If your audio system includes MIDI devices, you can also designate which MIDI input and output devices you want to use. For example, you can designate a MIDI keyboard to use for triggering commands and a MIDI synthesizer channel to use for playback. (See "About using MIDI devices" on page 39.)

To designate the devices you want to use:

1 Choose Options > Device Order.

2 Click the tab for the type of device you want to designate: Playback, Recording, MIDI Output, or MIDI Input.

3 Move the devices you want to use into the Multitrack Device Preference Order list by selecting devices in the Unused list and clicking Use. Remove the devices you don't want to use by selecting devices in the Multitrack Device Preference Order list and clicking Remove.

Note: *You can specify up to 16 stereo devices or 32 mono devices in the Multitrack Device Preference Order list.*

4 Designate the device you want to use in Edit View by selecting the device and clicking Use in EV. *[EV]* appears after the device name.

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ug.book Page 37 Tuesday, March 16, 2004 1:29 PM

5 Adjust the order of devices for use in Multitrack View by selecting a device and clicking Move Up or Move Down.

The first device in the list is the default device. This means that, by default, the first playback device is assigned as the output for all audio tracks in a session and the first recording device is assigned as the input for all audio tracks. Likewise, the first MIDI Out device is assigned as the output for all MIDI tracks. However, you can easily reassign the devices for a track. (See "Using the Track Properties window" on page 180.)

6 If desired, click a different tab to set up ordering for another type of device. When you are finished, click OK.

To quickly view or change the properties for a device, select the device and click Properties.

Setting properties for audio output devices

The Device Properties dialog box lets you specify Adobe Audition's parameters for playing back waveforms. If you have multiple sound cards, or a single card that has multiple audio outputs, you can customize the properties for each output.

To set properties for audio output devices:

1 Choose Options > Device Properties, and click the Wave Out tab.

2 Select a device from the list at the top of the dialog box.

The capabilities of the selected output device are shown in the Supported Formats table. A Yes or No indicates different combinations of sample rate and bit resolution. This table also shows what (if any) 32-bit formats the output device can handle, and whether it can accept the WDM driver extensible wave format.

3 Set any of the following properties. When you are finished, you can choose a different device to set up, or you can click OK to close the dialog box:

Order Displays the order of the device for use in Multitrack View. Click Change to open the Device Order dialog box and change the order of devices. (See "Designating which devices you want to use" on page 36.)

Use This Device In Edit View Indicates that Adobe Audition will use the device to play waveforms in Edit View.

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ug.book Page 38 Tuesday, March 16, 2004 1:29 PM

Limit Playback To Downsamples audio data for playback. Use this option to compensate for limitations imposed by your hardware. For example, if your sound card doesn't handle 32-bit audio correctly, you can have Adobe Audition limit the playback of 32-bit files to either 16-bit or 8-bit.

Send 32-bit Audio As Specifies how Adobe Audition sends 32-bit audio data to the output device. This option is not available if you select a Limit Playback To option. If the output device supports it, you can send 32-bit audio as 3-byte Packed PCM, 4-byte PCM, or 4-byte IEEE float.

Enable Dithering Activates dithering when playing back audio at a limited bit depth. If you deselect this option, Adobe Audition truncates the audio data instead. This means that bits that aren't used are simply chopped off and discarded. Enabling dithering is recommended when working with audio files that have a higher bit depth than your sound card supports. You can set the following options when dithering is enabled:

- bits specifies the number of bits to dither to. If you have a 20-bit sound card, for example, you will want to dither to 20 bits since any more bits will not be used by the card. Even for 16-bit-only sound cards, choosing to dither to 16-bit will improve the quality when playing back 32-bit audio.
- p.d.f. (probability distribution function) controls how the dithered noise is distributed away from the original audio sample value. Usually one of the Triangular p.d.f. functions is a wise choice, because it gives the best tradeoff between SNR, distortion, and noise modulation.
- Shaping specifies a noise shaping curve for moving noise to different frequencies. You can also specify that no noise shaping is used.

Setting properties for audio input devices

The Device Properties dialog box lets you specify Adobe Audition's parameters for recording waveforms. If you have multiple sound cards, or a single card that has multiple inputs, you can customize the properties for each audio input device.

To set properties for audio input devices:

1. Choose Options > Device Properties, and click the Wave In tab.
2. Select a device from the list at the top of the dialog box.

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ug.book Page 39 Tuesday, March 16, 2004 1:29 PM

The capabilities of the selected recording device are shown in the Supported Formats table. A Yes or No indicates different combinations of sample rate and bit resolution.

3 Set any of the following properties. When you are finished, you can choose a different device to set up, or you can click OK to close the dialog box:

Order Displays the order of the device for use in Multitrack View. Click Change to open the Device Order dialog box and change the order of devices. (See "Designating which devices you want to use" on page 36.)

Use This Device In Edit View Indicates that Adobe Audition will use the device to record waveforms in Edit View.

Get 32-bit Audio Specifies how the input device sends 32-bit audio data to Adobe Audition. If supported by the recording device, you can send 32-bit audio as 3-byte Packed PCM, 4-byte PCM, or 4-byte IEEE float.

Multitrack Latency Specifies the delay time (or *latency*) that the device introduces during recording. Many sound cards allow for monitoring input source signals with no latency. However, if you notice that tracks are out of sync, it is probably because one of the devices you used for recording introduced latency. Once you determine how out of sync a particular device gets, you can enter the number of milliseconds to delay a track's playback in relationship to all other tracks' playback to achieve synchronization.

Adjust To Zero-DC When Recording Removes any detected DC bias when recording.

About using MIDI devices

MIDI stands for Musical Instrument Digital Interface, and is a way of communicating performance information from one piece of software or hardware to another. This performance information can take the simple shape of a note instruction, as in E4, or it can transmit detailed information on things such as timing or sound patch data. Windows provides a way of transmitting MIDI information internally between programs, plus you can transmit MIDI information into and out of your computer to or from external devices (such as a MIDI Keyboard) through the MIDI port of a sound card, or other MIDI interface device.

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ug.book Page 40 Tuesday, March 16, 2004 1:29 PM

You cannot record audio directly from a MIDI input device into Adobe Audition. In order to work with MIDI data in Adobe Audition, you must save the MIDI data to a file using a MIDI sequencing application, and then import the MIDI file into a session as a clip. Once you have MIDI clips in a session, you can map them to a specific MIDI output device and channel for playback. (See "Working with MIDI tracks" on page 185.)

If you have a MIDI input device connected to your system's MIDI interface, you can use it to execute commands in Adobe Audition. For example, you can assign the Play command in Adobe Audition to the C4 note on your MIDI keyboard. This is called *MIDI triggering*. (See "Using shortcuts" on page 12.)

You can also use your system's MIDI Out and In ports to send and receive SMPTE/MTC timecode. This process lets you synchronize Adobe Audition's Multitrack playback and recording with other hardware or software components that also support SMPTE/MTC. (See "Using sessions as SMPTE masters or slaves" on page 166.)

Setting up for SMPTE synchronization

You can use SMPTE (Society of Motion Picture and Television Engineers) timecode to synchronize Adobe Audition's transport controls with a MIDI sequencing application or an external hardware device, such as a videotape machine. (See "Using sessions as SMPTE masters or slaves" on page 166.)

Adobe Audition sends and receives SMPTE timecode via MIDI timecode (MTC), which Windows transmits through your system's MIDI Out and MIDI In ports. MTC is a digital signal; to convert analog SMPTE timecode from a video or audio tape deck to digital MTC, you must use an appropriate MIDI interface.

To designate the devices with which you want to synchronize:

1 Choose Options > Device Properties.

2 Click the MIDI Out tab, and choose a device for SMPTE Output. This is the device to which Adobe Audition will send the MIDI timecode.

3 Click the MIDI In tab, and choose the device for SMPTE Slave Device. This is the device from which Adobe Audition will receive the MIDI timecode.

4 Click OK.

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ug.book Page 41 Tuesday, March 16, 2004 1:29 PM

To set options for incoming SMPTE timecode:

1 If your MIDI interface supports sample-accurate synchronization, choose Options > Sample Accurate Sync.

2 Choose Options > Settings, and click the SMPTE tab.

3 Set the following options:

Lead Time Specifies the amount of time (in milliseconds) in which Adobe Audition establishes synchronization with incoming timecode. Lower settings (200 and lower) result in faster transport response but may prevent Adobe Audition from establishing synchronization. Settings of 500 to 1000 are sufficient on most systems.

Stopping Time Specifies the amount of time (in milliseconds) Adobe Audition will continue playing if it encounters a dropout in timecode.

Lag Time Specifies the number of samples between incoming timecode and outgoing audio data. This value accounts for discrepancies introduced by sound card buffers. The default value is 10 samples.

Slack Specifies the number of frames Adobe Audition can fall out of sync with timecode before either repositioning the current-time indicator to match the code or performing a full resynchronization. A setting of up to 2.5 frames is recommended, as incorrect timecode is usually corrected on the next frame sent. The default value is 1 frame.

Clock Drift Correction Time Specifies the number of samples to crossfade when making time corrections to chase audio to timecode. The default value is 200 samples.

Reposition Playback Cursor When Shuttling Readjusts the playback position if synchronization is off by the Slack value.

Full Re-Sync When Shuttling Performs a full re-synchronization if synchronization is off by the Slack value.

Setting up external controllers

You can use external controllers, such as the Mackie Control, when recording and mixing in Adobe Audition. These devices let you edit audio tracks using real knobs and automated faders, instead of your mouse and computer keyboard. The Device Properties dialog box lets you configure external controllers and specify a volume increment.

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ug.book Page 42 Tuesday, March 16, 2004 1:29 PM

To set up external controllers:

1 Choose Options > Device Properties, and click the Ext. Controller tab.

2 Select the external controller you want to use, and specify a volume increment for the device.

3 Click Configure to set additional options for the device. These options are provided by the controller software. Refer to your controller documentation for more information.

4 Click OK.

Setting up ReWire connections

ReWire (a product of Propellerhead Software) is a technology for synchronizing audio applications. You can configure Adobe Audition to accept audio input from any ReWire-compatible application. When Adobe Audition is configured to accept ReWire input, it is referred to as a ReWire *host*. Applications that supply audio input are called ReWire *slaves* and the output channels they expose to the host are called devices.

To establish a ReWire connection, you first enable ReWire support in Adobe Audition and then activate a ReWire slave application and assign output from the slave to one or more Audition tracks. Adobe Audition serves as a ReWire host until you close the application. You can also manually disable ReWire support. For more information on using Adobe Audition as a ReWire host, see "Working with ReWire tracks" on page 184.

Note: *Before enabling ReWire in Adobe Audition you must close all other ReWire host and slave applications. After activating a slave application from within Adobe Audition, you will launch the application to establish the ReWire connection.*

To establish a ReWire connection:

1 In Multitrack view, choose Options > Device Properties and select the ReWire tab.

2 Click Enable. The dialog box automatically populates with a list of installed ReWire slave applications.

3 Select the check box next to the application you want to activate as a slave.

4 Choose one of the following track assignment options:

- Insert Summed Stereo Output Into First Available Track. All ReWire devices offer one summed stereo output. This option routes the summed stereo output into the first unoccupied track in the current session.

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ug.book Page 43 Tuesday, March 16, 2004 1:29 PM

- Insert All Outputs To Individual Tracks. ReWire devices may offer multiple channel outputs. This option routes each available ReWire output to its own track, starting with the first unoccupied track and following contiguously to additional unoccupied tracks.
- Insert Outputs Manually Using Track Device Input Dialogs. Choose this option if you want to assign outputs manually by using the Input Device dialog box. (See "Working with ReWire tracks" on page 184.)

5 Click Launch to launch the ReWire slave application and establish the ReWire connection. Adobe Audition assigns output from the ReWire slave to one or more tracks, as specified by the track assignment option you selected.

6 Open the session you want to work with in the ReWire slave application to make the audio available to Adobe Audition.

Note: *Because only one ReWire host can be active at a time, you need to disable ReWire in Adobe Audition before enabling any other ReWire host application.*

To disable ReWire support:

1 In Multitrack view, choose Options > Device Properties and select the ReWire Devices tab.

2 Click Disable, and then click OK.

Setting Adobe Audition preferences

The Settings dialog box lets you customize Adobe Audition's workspace, use of memory and hard disk space, spectral view, behavior when pasting, and other miscellaneous settings.

To use the Settings dialog box:

1 Choose Options > Settings.

2 Click a tab at the top of the dialog box to view the desired sets of options.

3 When you're finished setting options, click OK. To close the Settings dialog box without changing any options, click Cancel.

Once you click OK, most changes take effect immediately. If a change requires that you close and reopen Adobe Audition, you'll be prompted to do so. For example, you need to close and reopen Adobe Audition when you set up a different temporary folder.

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ug.book Page 44 Tuesday, March 16, 2004 1:29 PM

General options

The General tab in the Settings dialog box provides options for adjusting mouse behavior in Adobe Audition, as well as parameters for auto-play, live update, auto-scroll, and more.

Force Spacebar To Always Trigger Play Forces the spacebar to always trigger playback regardless of which dockable window has focus.

Auto-Play On Command-Line Load Enables the ability to start Adobe Audition and play a file from the command line. For example, if you go to the Run command in the Windows Start menu and type **"c:\Program Files\Adobe\Audition 1.5\Audition.exe" "c:\Program Files\Adobe\Audition 1.5\Audition Theme\TalkBackVerb.cel"** at the command line, Adobe Audition will start and begin playing TalkBackVerb.cel.

Live Update During Recording Enables live waveform drawing while recording. On faster computers, you can have the waveform displayed in real time as audio is being recorded. However, if you find the recorded audio becoming choppy, disable this option.

In Edit View's Spectral View mode, and at lower spectral resolutions (around 256), you can perform a nice scrolling spectral plot while recording with this option on.

Auto-Scroll During Playback And Recording Enables scrolling of the waveform display in sync with playback. Auto-scrolling only takes affect when you are zoomed in on a portion of a waveform and play past the viewed portion.

Note: *The display refresh rate is directly related to the Total Buffer Size setting in the System tab of the Settings dialog box. A low buffer size (such as 1) results in a smooth scrolling display, where as a high buffer size (such as 8) results in a more choppy display. (See "System options" on page 45.)*

Upon A Manual Scroll/Zoom/Selection Change Determines auto-scrolling behavior when a manual scroll, a zoom, or a selection change occurs in Adobe Audition. You can abort auto-scrolling until the next time you play or record; resume auto-scrolling only when the play cursor enters the view; or resume auto-scrolling immediately. Choose the one that best suits your needs.

Custom Time Code Display Defines the number of frames per second (FPS) assigned to the Custom time format in the View > Display Time Format menu.

Restore Default Workspace Resets all window sizes and positions to Adobe Audition's default arrangement.

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ug.book Page 45 Tuesday, March 16, 2004 1:29 PM

Ctrl Key Allows Dockable Windows To Dock Disables the Ctrl key from preventing a window to dock when moving the window around the work area.

Mouse Wheel Determines the amount to zoom in when rotating the mouse wheel found on Intellipoint-compatible pointing devices. Values from 10% to 80% work well. The higher the value, the further you'll zoom in when you roll the mouse wheel.

Time Selection Mouse Cursor Determines whether you want your mouse pointer to appear as an arrow or as an I-beam when it's over the waveform display.

Edit View Right-Clicks Determines the behavior for a right-click in the waveform display.

- Popup Menu: When right-clicking in the waveform display, a menu pops up if this option is selected. You can then Shift-click to extend a selection.
- Extend Selection: If you select this option, right-clicking in the Edit View's waveform display lets you extend the edge of a waveform selection instead of displaying the pop-up menu. To see the pop-up menu, hold down the Ctrl key as you right-click.

Default Selection Range Determines the amount of waveform data that automatically gets selected (if nothing is already highlighted) when you apply an effect.

- View: If this option is selected, the area that's automatically selected is limited to the area you can currently see on-screen.
- Entire Wave: When you choose this option, the entire waveform is automatically selected, even if you're only viewing a portion of it.

Double-clicking always selects the current view. Triple-clicking always selects the entire waveform.

Highlight After Paste Highlights the inserted selection when performing a Paste operation. Deselect this option to have the cursor placed at the end of the pasted selection instead.

Deselect this option for easier multiple pastes, one after the other.

System options

The System tab in the Settings dialog box provides options for configuring how Adobe Audition interacts with your system.

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ug.book Page 46 Tuesday, March 16, 2004 1:29 PM

Edit View Play/Record Buffer Determines the buffer size (in seconds) to be used when sending data to and from your sound card when playing back or recording in Edit View. Different sound card devices may require different memory buffer settings. The default settings should work fine for most sound cards, but if you hear choppiness (skips or dropouts) in recording or playback, you may need to adjust the buffer size or number of buffers used. For example, if you experience breakups in your audio, or you can't stop a recording in progress, increase the buffer size.

Use the two fields in the Edit View Play/Record Buffer area to reserve more memory for recording and playback by entering a higher buffer size, both in seconds and a number of buffers.

Keep in mind that while a greater buffer size will allow for increased multitasking when audio is being played, it does so at the expense of taking more of your computer's memory.

Wave Cache Determines the amount of memory that Adobe Audition reserves for processing data. Recommended cache sizes are from 8192 to 32768 KB (8192 KB is the default).

Select Use System's Cache to let Windows handle all disk caching. Keep in mind that Adobe Audition usually handles caching better than Windows can. However, this option reserves the least amount of memory, so it may be desired for systems with low amounts of RAM.

EV Preview Buffer Determines the minimum buffer size used when sending data to your sound card for the real-time Preview feature found in many effect dialog boxes. The default value is 250 milliseconds.

Different sound card ports may require different memory buffer settings. If you hear choppiness (skips or dropouts) when you use the Preview feature, try adjusting the buffer size used. (Choppiness can be caused by insufficient processing power as well.) Keep in mind that a larger Minimum Preview Buffer Size requires more computer memory.

Use Sound Card Positioning Info Allows Adobe Audition to query the sound card for the actual location and sync up the cursor with audio. This option is useful if a sound card doesn't play or record at 44,100 Hz (some sound cards, for example, work at 44,050 Hz or 44,130 Hz). Leave this option unselected unless the cursor is out-of-sync with the audio.

CD Device Options Specifies the SCSI interface used by your CD device: ASPI (Advanced SCSI Programming Interface) or SPTI (SCSI Pass Through Interface).

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ug.book Page 47 Tuesday, March 16, 2004 1:29 PM

Temporary Folders Specifies the folders in which you want Adobe Audition to store temporary files. Adobe Audition creates temporary files for use when performing edits on your audio. All temporary files begin with CEP and have the .tmp extension. The rest of the filename is chosen at random when the file is created. If there are no copies of Adobe Audition running, none of these files should be present, since Adobe Audition normally deletes temporary files when it exits. However, these files can be left behind in extreme circumstances if Adobe Audition crashes, or if Windows unexpectedly quits while Adobe Audition is active. As long as Adobe Audition isn't running, you can safely delete these files. You can also use the Manage Temporary Folder Reserve Space to delete temporary files you aren't using while Adobe Audition is running. (See "Managing temporary files" on page 57.)

Important: *You need to have enough space available in these folders to accommodate the total size of all the audio files you wish to edit simultaneously.*

Use the reserve free fields to specify an amount to leave available for headroom purposes for both the primary and secondary temporary folders.

- Temp Folder: Specifies Adobe Audition's main temporary folder. Ideally it should be on your fastest hard drive.
- Secondary Temp: Specifies Adobe Audition's secondary temporary folder. For best results, this should be on a different physical hard drive than the primary temp folder. This is especially true when recording more than one track at a time in Multitrack View, because odd track recordings go to one temp folder while the even tracks are recorded to the other temp folder, dividing the workload.

Note: *Providing you have enough free space on the drive that holds the primary temporary folder, Adobe Audition will work just fine if no Secondary Temp folder is specified.*

Undo Specifies options for Adobe Audition's Undo feature, which lets you revert back to your last edit with a keyboard shortcut (Ctrl+Z), menu command, or toolbar button.

- Enable Undo: Activates the Undo function. Because Undo requires extra disk space for its temporary files and time to save them before processing, you may sometimes want to turn this feature off.
- Levels (minimum): Specifies the fewest number of Undo levels.
- Purge Undo: Deletes all of Adobe Audition's Undo files. This frees up disk space, but ends your ability to revert to previous edits.

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ug.book Page 48 Tuesday, March 16, 2004 1:29 PM

Delete Clipboard Files On Exit Deletes Adobe Audition clipboard files when you exit. In general, leave this option enabled: Usually, after you finish with an Adobe Audition session, these clipboard files are no longer needed and just take up valuable hard disk space.

Deselect this option to retain Adobe Audition's clipboard files on your hard drive after you exit the program.

Force Complete Flush Before Saving Disables the quick save feature, in which Adobe Audition quickly saves files that contain only minor modifications. If you enable this option and force a flush before saving, Adobe Audition saves all files by making a backup copy of the file internally and then writing the entire file back.

This option is disabled by default. When enabled, it considerably increases the save time for large files. It is intended for use only if you have trouble saving back to the same filename or you have a problem with Adobe Audition's quick save feature.

Colors options

The Colors tab in the Settings dialog box provides options for changing Adobe Audition's color scheme.

Color Presets Lists color scheme presets that come with the program as well as those you've created yourself. To choose one, select it from the list. The currently selected color scheme is displayed in the Example window.

Save As Saves the currently selected color scheme as a preset.

Delete Deletes the currently highlighted color scheme preset.

Waveform Tab Lists all of Adobe Audition's waveform elements to which you can assign custom colors. Choose an item from the list and click the Change Color button to change the color.

To adjust the appearance of the selected (highlighted) portions of waveforms and blocks, select a Selection option:

- Transparency: Drag the slider or enter a value to adjust the transparent value (in percentage) of a selection; 0 is no transparency and 100 is maximum transparency.
- Invert: Select to set the selection colors to the inverse of the nonselected colors.

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ug.book Page 49 Tuesday, March 16, 2004 1:29 PM

Spectral Tab Lists the display elements for Adobe Audition's spectral display. Select an item from the list and click the Change Color button to adjust the element's color.

For Spectrum, choose one of these options:

- Reverse Direction: Inverts the normal colors of the spectrum display, similar to an Invert or Negative command in a photo editor.
- Gamma: Adjusts the overall brightness of the Spectral View. Positive numbers make the display brighter, while negative numbers darken the display. This setting works just like the Gamma function in many image editors.

To adjust the appearance of the selected (highlighted) portions of waveforms in spectral display, choose a Selection option.

- Transparency: Drag the slider or enter a value to adjust the transparent value (in percentage) of a selection; 0 is no transparency, and 100 is maximum transparency.
- Invert: Select to set the selection colors to the inverse of the nonselected colors.

Controls Tab Lists the Adobe Audition control elements for which you can change colors. Select an item from the list and click the Change Color button to adjust the element's color.

Select Segmented Progress Bar to make the progress bar segmented instead of solid. The progress bar appears when you apply an effect, or open or save large waveforms.

Select White Progress Background to make the background of the progress bar white.

For Dockable Windows, select one of the following options:

- Use System 3D Color: Select to make dockable windows use your system's 3D color. This is the color Windows uses to render most windows on your system.
- Use Darkened System 3D Color: Select to make dockable windows use the darkened version of your system's 3D color.
- Use Specified 3D Color: Select to make Adobe Audition's dockable windows use the 3D color you specify.

To change the 3D color, select Dockable Windows 3D Color in the controls list, and then click the Change Color button to select a new color.

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ug.book Page 50 Tuesday, March 16, 2004 1:29 PM

Display options

The Display tab in the Settings dialog box provides options for adjusting Adobe Audition's Spectral View and Waveform View modes.

Windowing Function Determines the method Adobe Audition uses to segment the spectral data before it displays it. The segments (windows) are listed in order from the narrowest frequency band/most noise to the widest frequency band/least noise. Blackmann or Blackmann-Harris are good choices.

Resolution Specifies the number of vertical bands used in drawing frequencies. Keep in mind that the larger this number, the longer it will take for Adobe Audition to render the spectral display. Performance will vary based on the speed of your computer.

Window Width Specifies the width of the window (or frame size) used in plotting the spectral data, where 100% is a frame size of the FFT (Fast Fourier Transform) size. Window Width basically lets you increase time resolution at the expense of some frequency resolution. So the display will become more accurate along the timeline (left and right) and less accurate along the frequency scale (up and down) as the window width decreases. The default setting is 75%, but you should lower the value (50 to 75% works best) if you want to increase the resolution horizontally—for example, to find out exactly where a certain frequency starts.

Plot Style Specifies a style for plotting frequencies:

- Logarithmic Energy Plot: In this mode, colors change with the decibel value of the energy at any particular time and frequency. In this mode, you can see more details in the very quiet ranges, especially if the Range is quite high (above 150 dB). Use the Range value to adjust the sensitivity in plotting frequencies.
- Linear Energy Plot: When selected, colors are chosen based on percentage of maximum amplitude instead of decibel amplitude. Liner Energy Plot can be useful for viewing the general overview of a signal without getting bogged down by detail at much quieter levels. You can adjust the Scaling factor to highlight audio of different intensities.

Show Cue And Range Lines Displays cue marker and range lines in the waveform display. Cue marker and range entries in the Cue List appear with vertical dotted lines overlaying the audio, connecting the arrows from the top to the bottom of the display.

Show Grid Lines Displays grid lines in the waveform display. The grid lines mark off time on the horizontal *x*-axis and amplitude on the vertical *y*-axis.

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ug.book Page 51 Tuesday, March 16, 2004 1:29 PM

Show Center Lines Displays center lines in the waveform display. The center lines represent zero amplitude of the waveform's right and left channels.

Show Boundary Lines Displays boundary lines in the waveform display. Boundary lines are the horizontal lines that visually indicate where the waveform's amplitude approaches or exceeds the clipping level. The value in the Display Boundary Lines At option specifies the amplitude at which the boundary lines appear.

Peak Files Specifies options for peak (.pk) files, which Adobe Audition uses to store information about how to display WAV files. Peak files make file opening almost instantaneous by greatly reducing the time it takes to draw the waveform (especially with larger files).

- Peaks Cache: Determines the number of samples per block to be used when storing peak files. Larger values reduce the RAM requirement for large files at the expense of slightly slower drawing at some zoom levels. If RAM is an issue on your system, and you're working with very large files (several hundred megabytes or more in size), consider increasing the Peaks Cache to 1024 or even 1536 or 2048.
- Save Peak Cache Files: Specifies that peak files are saved with all .wav files (in the same folder) with the extension .pk following the original audio filename.
- Rebuild Wave Display Now: Click to rescan the current file for sample amplitudes and redraw the waveform.

Data options

The Data tab in the Settings dialog box provides options for controlling how Adobe Audition handles audio data.

Embed Project Link Data For Edit Original Functionality Links session files with exported mixdown files. Once these files are linked, you can select a mixdown file in Adobe Premiere Pro or After Effects, and then open and remix the related session in Adobe Audition's Multitrack View.

Auto-Convert All Data To 32-Bit Upon Opening Converts all 8-bit and 16-bit data to 32-bit when a file is opened, and all subsequent operations will keep the data in the 32-bit realm.

Interpret 32-Bit PCM .wav Files As 16.8 Float Causes this version of Adobe Audition to be compatible with previous versions when it comes to handling 32-bit PCM .wav files.

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ug.book Page 52 Tuesday, March 16, 2004 1:29 PM

Dither Transform Results (increases dynamic range) Enables dithering when processing effects such as FFT Filter or Amplify. Most processing done by Adobe Audition uses arithmetic greater than 16-bit, with the results converted back to 16-bit when complete. During this conversion, dithering provides a higher dynamic range and cleaner results, with less distortions and negative artifacts. With dithering, you get almost 24-bit sample performance in only 16 bits, as the dynamic range is increased by another 10 dB or so, allowing signals as quiet as –105 dB.

If this option is disabled, the results are truncated to 16 bits when converting back, thus losing the more subtle information.

When enabled, the addition of dither retains this subtle information. The drawback is that with each operation a small amount of white noise is added at the quietest volume level. However, the trade-off between using dither (thus adding noise) and truncating the data (thus creating artifacts and correlated quantization noise) generally favor using dither, so it's best to leave this option enabled.

Use Symmetric Dithering Enables symmetric dithering. In most cases, it's best to leave this option selected. If unselected, a DC offset of one-half sample is added each time data is dithered. Symmetric dithering has just as many samples added above zero as below zero. By contrast, nonsymmetric dithering just toggles between 0 and 1. Sometimes in a final dither, this may be desired to reduce the bit range of the dither. However, both methods produce identical audible results in every respect.

Smooth Delete And Cut Boundaries Smooths Cut and Delete operations at the splicing point, preventing audible clicks at these locations.

Smooth All Edit Boundaries By Crossfading Automatically applies a crossfade to the starting and ending boundaries of the selection. This option smooths any abrupt transitions at these endpoints, thus preventing audible clicks when filtering small portions of audio. You can enter a value (in milliseconds) in the crossfade time box to specify the crossfade duration to be applied.

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ug.book Page 53 Tuesday, March 16, 2004 1:29 PM

Auto-Convert Settings For Paste When pasting different sample formats, Adobe Audition uses these settings when auto-converting the clipboard to the current sample format. Valid settings range from 30 to 1000.

- Downsampling Quality Level: Enter a value (30 to 1000) for downsampling quality. Higher values retain more high frequencies while still preventing the aliasing of higher frequencies to lower ones. A lower quality setting requires less processing time, but results in certain high frequencies being rolled off, leading to muffled-sounding audio. Because the filter's cutoff slope is much steeper at higher quality settings, the chance of ringing at high frequencies is greater. Usually values between 80 and 400 do a great job for most conversion needs. The default value is 80.
- Pre-Filter: To prevent any chance of aliasing, the pre-filter on downsampling removes all frequencies above the Nyquist limit, thus keeping them from generating false frequencies at the low end of the spectrum. In general, select this option for best results.
- Upsampling Quality Level: Enter a value (30 to 1000) for upsampling quality. Higher values retain more high frequencies while still preventing the aliasing of higher frequencies to lower ones. A lower quality setting requires less processing time but results in certain high frequencies being rolled off, leading to muffled-sounding audio. Because the filter's cutoff slope is much steeper at higher quality settings, the chance of ringing at high frequencies is greater. Usually values between 100 and 400 do a great job for most conversion needs. The default value is 120.

You should use a higher value whenever you downsample from a high sample rate to a low rate. For upsampling, a lower value produces quality almost identical to a higher value. The difference lies in the larger phase shift that exists at higher frequencies, but since the phase shift is completely linear, it's very difficult to notice. Downsampling, at even the lowest values, generally doesn't introduce any undesired noisy artifacts. Instead, the sound might be slightly muffled because of the increased high-end filtering.

- Post-Filter: To prevent any chance of aliasing, the post-filter on upsampling removes all frequencies above the Nyquist limit, thus keeping them from generating false frequencies at the low end of the spectrum. In general, select this option for best results.

Dither Amount For Saving 32-Bit Data To 16-Bit Files Enables dithering when pasting 32-bit audio to 16-bit. The default value of 1 (bit) enables dithering, while a value of 0 disables dithering. For semi-dithering, choose a value of 0.5.

With dithering, you get almost 24-bit sample performance in only 16 bits, as the dynamic range is increased by another 10 dB or so. This allows signals as quiet as –105 dB.

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ug.book Page 54 Tuesday, March 16, 2004 1:29 PM

Allow For Partially Processed Data After Canceling Effect Determines what happens after you click the Cancel button while in the middle of applying an effect to a waveform. When selected, Adobe Audition leaves the effect applied to all data processed up until the point you clicked Cancel. When deselected, Adobe Audition automatically removes the effect on already processed data when you click Cancel.

Multitrack options

The Multitrack tab in the Settings dialog box provides options that let you optimize performance during recording, playback, and mixdown.

Playback Buffer Size Determines the buffer size (in seconds) used when sending data to your sound card when playing back a multitrack session. Different sound card drivers may require different memory buffer size settings. Adobe Audition's default settings should work fine for most sound cards. If you hear choppiness (skips or dropouts) in multitrack playback, adjust the buffer size. (Choppiness in multitrack playback can also be attributed to the background mixing process not being far enough ahead). A larger buffer size requires more computer memory. The default setting is 1.

Playback Buffers Specifies the number of buffers Adobe Audition uses for playback in the multitrack environment. If you experience break-up in your audio, try reducing the number of buffers. Increasing this number might also be helpful for some configurations. The default setting is 10.

Recording Buffer Size Reserves memory for recording in a multitrack session by entering a buffer size (in seconds). Different sound card drivers may require different memory buffer size settings. Adobe Audition's default settings should work fine for most sound cards. If you experience dropouts while recording in multitrack (especially when playback seems fine), try increasing this setting. (First, be sure the background mixing process is sufficiently complete when you go to record as this may cause the same symptom.) A larger buffer size requires more computer memory. The default setting is 2 seconds.

Recording Buffers Specifies the number of buffers used for recording in the multitrack environment. If you experience break-ups in your audio, try reducing the number of buffers. Increasing this number may also help for some configurations. The default setting is 10.

Background Mixing Priority Specifies the priority level of the background mixing process in a multitrack session. Lower values indicate a higher level of priority above other system events. You can use fractional numbers (such as 0.8). The default setting is 2.

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ug.book Page 55 Tuesday, March 16, 2004 1:29 PM

Open Order Determines the order in which Adobe Audition opens a sound card's playback (in) and record (out) ports for use in the multitrack environment. This order is relevant only for older sound cards that don't support full-duplex capability.

Start Order Determines the order in which Adobe Audition starts a sound card's playback (in) and record (out) ports for use in the multitrack environment. This order is relevant only for older sound cards that don't support full-duplex capability.

Correct For Drift In Recordings Synchronizes the master audio playback device (generally, the first Out device listed in the session—the one on Track 1) and the record device of the waveform being recorded. If the true sample rates on the cards differ enough that the recording would have drifted out of sync with the original if both were played back at exactly the same sample rate, then the recording is corrected by resampling to make it the proper length. This option only works with new record tracks, not with recording on top of existing waveforms, or punch-ins.

Note: *On sound cards that support sample accurate devices (that is, synchronized device starting, and all devices keyed off of the same clock) you don't need to select this option. This option allows for some measure of near sample-accurate synchronization across different sound cards, or when using with a single sound card that doesn't use the same clock for playback and recording (which is common in consumer and other low-end sound cards).*

Correct For Start Sync In Recordings Compares the exact true time that the record device started with the time the master playback device started. If different, the recorded block's position is adjusted so the recording starts in perfect sync with the playback. This option only works with new record tracks, not with recording on top of existing waveforms, or punch-ins.

If this option is enabled, and you do a loopback test (by connecting the audio Out to the audio In and recording some ticks) and each recording is still a fixed amount out of sync, then you can adjust for this by entering this amount (in milliseconds) in the Latency field of Options > Device Properties for the recording device being used. To compute milliseconds, look at the difference in samples, multiply by 1000, and then divide by the sample rate. For instance, if the recording consistently appears 27 samples ahead of the playback, the latency would be 27 x 1000 / 44,100, or about 0.61 milliseconds. (The reason for the milliseconds format and not samples is because at various sample rates this latency will be different in terms of samples, but will be the same in terms of milliseconds.)

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ug.book Page 56 Tuesday, March 16, 2004 1:29 PM

Note: *On sound cards that support sample accurate devices (that is, synchronized device starting, and all devices keyed off of the same clock) you don't need to select this option. This option allows for some measure of near sample-accurate synchronization across different sound cards or a situation where a single sound card uses different clocks for playback and recording. (This situation is common for consumer and other low-end sound cards.)*

Delete Old Takes After Merging Automatically deletes any unused takes created during a punch-in when you select a take. If you don't select this option, unused takes remain available to the Session (in the Insert menu) and occupy hard drive space.

Crossfade Time Determines the amount of time (in milliseconds) over which crossfading occurs when a take created using punch-in is merged back into the surrounding waveform.

Mixdowns Determines the bit-resolution that is used when performing a mixdown. Regardless of the session format (16-bit or 32-bit), you can generate mixdowns at either 16-bit or 32-bit quality with this option. The default is 16-bit. Click Dithering Option to specify how to dither the 16-bit mixdown.

Track Record Specifies how waveforms are created when recording directly into the Multitrack View: as mono or stereo, and as 16-bit or 32-bit.

Pre-Mixing Determines the bit size used for the background mixing process. Best quality is achieved by leaving this at the default 32-bit setting. However, if you're using multiple sound cards, it may be advantageous and faster to choose 16-bit for pre-mixing, as less data will be transferred across the hard drives. For single output device situations, or faster hard drives, 32-bit is better as it provides optimization at mixdown.

Panning Mode Specifies the method used for panning waveforms in a multitrack session.

- L/R Cut (log): Pans left by reducing the volume of the right channel, and pans right by reducing the left channel volume. The channel being panned to doesn't increase in volume as panning gets closer to 100%.
- Equal-power Sine: Pans left and right channels with equal power, so a hard pan left will contain the same loudness as both channels together. This results in an increase of 3 dB RMS on the channel being panned to when at 100%.

Note: *Because panning can actually make one channel louder than the original waveform, audible clipping can occur in 16-bit sessions. To avoid this, work in the 32-bit realm if you're using the Equal-power Sine panning method.*

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ug.book Page 57 Tuesday, March 16, 2004 1:29 PM

Auto Zero-Cross Edits Automatically adjusts the beginning and end points of all Cut, Copy, and Paste-type edits to the nearest place where the waveform crosses the center line (zero amplitude point).

If the amplitudes aren't lined up on both sides of the selection, the endpoints are at different amplitudes. This often results in an audible pop or click at that point.

Smooth Auto-Scrolling During Playback Enables smooth scrolling when playing back audio in Multitrack View. By default Adobe Audition uses a paging method of scrolling in Multitrack View instead of the smooth scrolling technique used in Edit View. This saves on system resources.

Save Locked Track Files After Closing Sessions Saves the temporary files associated with locked tracks. When you reopen the session, Adobe Audition uses the temporary file instead of mixing down the locked tracks.

SMTPE options

The SMTPE tab in the Settings dialog box provides options for adjusting the settings for incoming SMTPE timecode. For more information, see "Setting up for SMPTE synchronization" on page 40.

Managing temporary files

When you edit a file, Adobe Audition converts the audio data into an internal, temporary waveform. This process allows for quicker editing, better handling of large files, and the ability to undo changes. You can specify the folders where you want Adobe Audition to save temporary files and customize Undo options in the System tab of the Settings dialog box. (See "Setting Adobe Audition preferences" on page 43.)

One advantage to using temporary files is virtually unlimited waveform sizes, since the maximum waveform size depends only on the size of your hard drive. The drawback, of course, is that the temporary file can get extremely large, potentially preventing you from being able to save a masterpiece on the same drive. If you notice long delays between edits or stuttering sounds on playback, you may be running out of free disk space in which to save the temporary file. In this case, you can use the Manage Temporary Folder Reserve Space dialog box to delete temporary files you're not using, clear specific Undo items, and change the amount of reserve space. This dialog box automatically appears when available hard drive space nears zero kilobytes.

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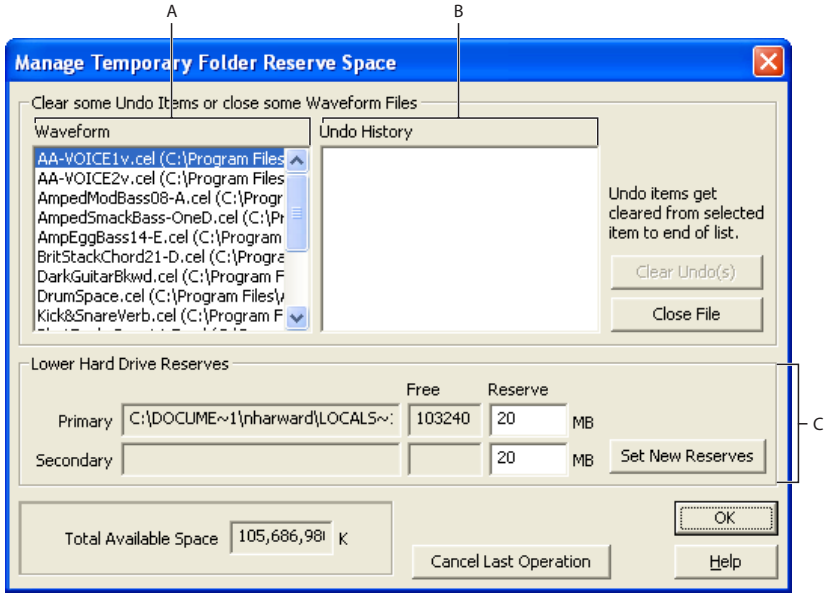
ug.book Page 58 Tuesday, March 16, 2004 1:29 PM



Manage Temporary Folder Reserve Space dialog box
A. *Open waveforms* ***B.*** *Undo items for the selected waveform* ***C.*** *Location of primary and secondary temporary folders*

Use the Status Bar to monitor the amount of free disk space. (See "Using the status bar" on page 21).

Adobe Audition doesn't create a temporary file for a waveform until you edit the waveform. However, you can force Adobe Audition to create a temporary file by using the Flush Virtual File command. This is useful when you need to use a waveform simultaneously in Adobe Audition and another application.

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ug.book Page 59 Tuesday, March 16, 2004 1:29 PM

To manage temporary folder reserve space:

1 Choose File > Manage Temporary Folder Reserve Space.

2 Do any of the following:

- To close a temporary file you're no longer using, select the file in the Waveform list, and click Close File. (The currently active waveform can't be closed this way, however.)
- To clear Undo items for a file, select the file in the Waveform list. The Undo History list displays the actions that are currently being retained on your system and the amount of hard drive space each instance consumes. Select an item and click Clear Undo(s). All items at the selected level and below are removed.
- To change the amount of space you want to keep free on the drives where the temporary files reside, enter a value in the Reserve text box, and click Set New Reserves.
- To stop any action in progress, such as the application of an effect or any other edit, click Cancel Last Operation. This option is useful only if the dialog box automatically appeared because you ran out of storage space.

If Adobe Audition crashes, there may be a temp file (CEPx.tmp) in your temporary folder that you should manually delete.*

To force Adobe Audition to create a temporary file for the current waveform:

In Edit View, choose File > Flush Virtual File.

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ug.book Page 61 Tuesday, March 16, 2004 1:29 PM

Chapter 3: Importing, Recording, and Playing Audio

You can bring audio into Adobe Audition by importing it from an audio or video file or by recording it from an external source, such as a microphone or a computer's CD player. When you play back the audio, Adobe Audition provides an assortment of features for monitoring the sound.

Opening audio files and multitrack sessions

Both Edit View and Multitrack View provide a variety of methods for opening files. In Edit View, you can open audio files; in Multitrack View, you can open session files.

Opening audio files in Edit View

In Edit View, you can open audio from a variety of audio file formats, including MP3, WAV, and AIFF. For more information on supported file formats, see "Choosing an audio file format" on page 231.

If desired, you can change the sample type of the audio when you import it or append the audio to the end of the current waveform. Whichever method you choose for opening files, Adobe Audition provides options that let you preview the contents of files before you open them.

To open an audio file:

1 In Edit View, choose File > Open. Alternatively, click the Open button in the toolbar or the Import button in the Files tab of the Organizer window.

2 Locate and select the file you want to open. To select multiple, adjacent files, click the first file and Shift-click the last. To select multiple, nonadjacent files, Ctrl-click them.

Note: *If you don't see the name of the file you want, choose All Supported Media from the Files Of Type menu. If you still don't see the file, it might be stored in a format that Adobe Audition can't read.*

3 Click Open.

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ug.book Page 62 Tuesday, March 16, 2004 1:29 PM

To preview the contents of a selected file:

Do any of the following:

- Click Play to listen to the file once.
- Select Loop to repeat the file until you click Stop.
- Click Auto Play to play files automatically when you select them.

To append an audio file to the current waveform:

1 In Edit View, choose File > Open Append.

If the new audio has a different sample rate, resolution, or channel type than the current waveform, Adobe Audition converts it to match the current waveform. For the best results, append files that have the same sample rate as the waveform.

2 Locate and select the file you want to open. To select multiple, adjacent files, click the first file and then Shift-click the last. To select multiple, nonadjacent files, Ctrl-click them.

3 Click Open.

To convert audio to a different sample rate, resolution, or channel type during import:

1 In Edit View, choose File > Open As.

2 Locate and select the file you want to open, and click Open.

3 Set the desired options in the Open File(s) As dialog box, and click OK:

Sample Rate Determines how many frequencies can be encoded in the audio signal. (Higher sampling rates mean wider bandwidth.) For more information, see "About sample rates" on page 110.

Channels Determines if the waveform is mono or stereo. Select Mono to create a waveform with just one channel of audio information. This option works well for a voice-only recording. Select Stereo to create a two-channel waveform with separate right and left channels. This option is usually best for a music recording. Because they contain twice as much data, stereo waveforms consume twice the storage space of mono waveforms.

Resolution Determines the number of unique amplitude levels Adobe Audition can use to represent a sound. The 32-bit level is best while you work in Adobe Audition, and you convert down for output if necessary.

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ug.book Page 63 Tuesday, March 16, 2004 1:29 PM

Note: Older sound cards might not be able to play 32-bit files properly. To check the capabilities of your sound card, choose Options > Device Properties. If your sound card doesn't support 32-bit files, you can convert the files to a lower bit rate (such as 16-bit) for playback.

Opening session files in Multitrack View

Session files contain no audio data themselves. Instead, they are small files that point to other audio files on the hard drive. A session file keeps track of what files are a part of the session, where they go in the multitrack, what envelopes and effects are applied to the tracks, and so on. For more information on creating session files, see "Creating new sessions" on page 162.

In Multitrack View, you can open individual session files and you can append one session to another to quickly build elaborate compositions with shared themes. When you append sessions, appended tracks appear below current tracks. For example, if the current session uses tracks 1-4, Adobe Audition appends tracks 5 and greater and places them at the beginning of the timeline. If desired, you can then move clips in appended tracks to a new position. (See "Selecting and moving clips" on page 168.)

Note: You can append a session only if it uses the same sample rate and bit depth as the current session. The sample rate and bit depth of the current session is displayed in the status bar.

To open a session file:

1 In Multitrack View, choose File > Open Session. Alternatively, click the Open button in the toolbar.

2 Locate and select the file you want to open, and click Open.

To append a session file to the end of the current session:

1 In Multitrack View, choose File > Append To Session.

2 Locate and select the file you want to open, and click Open.

Inserting audio files into multitrack sessions

When you insert an audio file in Multitrack View, the file becomes an audio clip on the selected track. For more information about audio clips, see "Working with clips" on page 168.

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ug.book Page 64 Tuesday, March 16, 2004 1:29 PM

To insert an audio file into a multitrack session:

1 In Multitrack View, position the current-time indicator at the desired insertion point.

2 Select the desired track.

3 Do one of the following:

- Choose Insert > Audio, select the audio file, and click Open. To preview the contents of a selected file, click Play to listen to the file once, or click Auto Play to play the file automatically when you select it. Select Loop to repeat the file until you click Stop.
- Choose Insert, and select the name of a recently opened waveform from the submenu.
- Choose Insert > File/Cue List. A window appears that lists all of the files that are currently open in Edit View. If a file has cues in it, a plus sign (+) appears next to its name to let you expand that file and see all the cue ranges in it. Click the file or cue you want to insert. Alternatively, drag the file or cue into the track display.
- Select one or more files in the Files tab of the Organizer window, and click the Insert Into Multitrack button . If you select multiple files, each is inserted into a separate track. This method lets you insert a file into a session without leaving Edit View. (See "Organizing files" on page 24.)

Note: *If the audio file is longer than the space available on the selected track, Adobe Audition inserts the new clip on the nearest empty track.*

Importing audio from CD

If you want to import audio into Adobe Audition from a CD, you can digitally extract it or record it internally. Digital extraction is the recommended method because it produces higher-quality audio than internal recording. Only use internal recording if your CD-ROM drive doesn't support digital extraction.

Extracting tracks from CDs

If your computer's CD-ROM drive supports audio digital extraction (also known as *ripping*), you can extract tracks from audio CDs. Once the audio is in Adobe Audition, you can edit it like any other waveform. Of course, if the CD is a typical read-only compact disc, you won't be able to save those changes back to CD. Instead, save modified CD tracks to a hard disk or burn them onto a new CD.

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ug.book Page 65 Tuesday, March 16, 2004 1:29 PM

Adobe Audition provides two methods for ripping tracks from CDs: using the Open command and using the Extract Audio From CD command. Using the Open command is the quickest method and is preferred for ripping entire tracks. Using the Extract Audio From CD command gives you more control, such as the abilities to rip partial tracks and specify the ripping process used.

To extract tracks from a CD by using the Open command:

1 Place an audio CD in the computer's CD-ROM drive.

2 In Edit View, choose File > Open.

3 Choose CD Digital Audio (*.cda) as the file type, and navigate to the computer's CD-ROM drive.

4 Select the tracks you want to rip, and click Open.

To extract tracks from a CD by using the Extract Audio From CD command:

1 Place an audio CD in the computer's CD-ROM drive.

2 In Edit View or CD Project View, choose File > Extract Audio From CD.

3 For Device, choose the drive that contains the audio CD.

4 For Source Selection, do one of the following:

- Select Track to extract one or more complete CD tracks. A list of all tracks on the CD appears, along with their lengths stated in Min:Sec:Frame format. (Each second of CD audio has 75 frames.)
- Select Time to extract part of a track or a segment of audio that spans multiple tracks. Enter the beginning frame in the Start box, and the total number of frames you wish to extract in the Length box. (Each second of CD audio has 75 frames.) The actual start and length times appear in Min:Sec:Frame format above their respective boxes. The Range bar provides a graphical representation of how much audio will be extracted and where the audio appears within the CD. However, if you select only a short bit of audio to extract, you might not see any change in the Range bar.

The Time option is great for pulling hidden tracks from CDs, as well as for joining tracks that have been broken up by track indexes (such as performance track CDs and live albums).

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ug.book Page 66 Tuesday, March 16, 2004 1:29 PM

5 For Interface Option, choose Generic Win32 or ASPI/SPTI. In most cases, ASPI/SPTI is the best choice. Select Generic Win32 only if the ASPI/SPTI option doesn't produce satisfactory results. The Generic Win32 option causes the Extract Audio From CD feature to use Input/Output control codes instead of SCSI commands.

For more information, see "Extract Audio From CD options" in Help.

6 For Error Correction, CDDA Accurate is automatically selected if the CD-ROM drive has built-in ripping error correction. For these types of drives, no error correction is needed, so you won't be able to select any options from this part of the Extract Audio From CD dialog box.

However, if your drive isn't CDDA Accurate, you have access to No Correction and Jitter Correction options. No Correction, as you'd expect, means that no error correction will be performed. Jitter Correction compensates for data reading problems that older drives might have.

7 To listen to the selected tracks before extracting them, click Preview.

8 To save the settings for future use, save a preset. (See "Using presets" on page 28.)

9 After you finish setting options, click OK.

Extract Audio From CD options

If you select ASPI/SPTI in the Extract Audio from CD dialog box, set the following options as desired:

Read Method Lets you choose the way Adobe Audition reads CD audio. Several methods are provided, many of them developed before the SCSI 3 specifications were published. (The SCSI 2 specs don't accommodate CD ripping.)

- MMC – Read CD is a SCSI 3-specific setting, and it works with most all recent drives. If you have a newer CD-ROM drive, try this setting first.
- SBC – Read10 is a standard SCSI read setting that uses a 10-byte SRB (SCSI Request Block). All SCSI devices are required to support this setting.
- SBC – Read6 is a standard SCSI read setting that uses a 6-byte SRB (SCSI Request Block). Many SCSI devices support this setting, but because it's optional, not all do.
- Plextor (D8) sends the D8 SCSI Op Code to the CD-ROM drive. Use this setting with older Plextor CD-ROM drives.

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ug.book Page 67 Tuesday, March 16, 2004 1:29 PM

- D5 sends the D5 SCSI Op Code to the CD-ROM drive.
- NEC works with older NEC CD-ROM drives.

CD Speed Lists all extraction speeds that the selected CD-ROM drive supports and lets you specify the speed you want to use. The Max (Maximum) Speed option usually produces satisfactory results, but if it produces errors, specify a slower speed.

Buffer Size Specifies how much data Adobe Audition calls into the CD Extraction module to fetch, therefore determining how much data is pulled from the CD in each call to the read command. The default is 16 KB, but you can experiment with other sizes (which range all the way to the highest buffer size the CD-ROM drive supports). Although higher sizes mean faster ripping, they *could* introduce errors into the ripped file.

Swap Byte Order Changes the byte order from Little Endian to Big Endian, or vice-versa. Some CD-ROM drives designed to work only with other types of computers (like DEC and Macintosh systems) report data by using the Little Endian byte order, while PCs use the Big Endian method. Normally, you should leave this box unchecked; check it *only* if the extraction process seems to work fine but the audio results are "garbage."

Swap Channels Places the left channel of a CD's audio in the right channel of the Wave Display, and places the right channel of the audio in the Wave Display's left channel.

Spin Up Before Extraction Causes the CD-ROM drive to start spinning before Adobe Audition extracts the data. Some CD-ROM drives have better accuracy if they first read the CD after the drive is spinning. Selecting this option for other drives, however, doesn't provide any advantage.

Recording from CDs internally

If you have an older CD-ROM drive that doesn't support digital extraction, or if you have problems ripping a track into Adobe Audition, then you can record from a CD in real-time through the sound card on your computer. This method is called *internal recording*. Keep in mind that not all PC's have an analog cable from a CD drive, and not all computers react the same way when recording from CD internally. As a result, this method is never preferable to extracting from CD digitally.

Before you record from a CD internally, you should always preview the CD Audio input level to make sure that clipping won't occur.

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ug.book Page 68 Tuesday, March 16, 2004 1:29 PM

To preview the CD Audio input level:

1 Open your favorite third-party CD player application (such as Windows Media Player).

2 Start playing the loudest part of the CD. Then, switch to Adobe Audition, and choose Options > Monitor Record Level.

3 Use the Level Meters in Adobe Audition to monitor the amplitude of the incoming signal. You want the input level to be as loud as possible without exceeding 0 dB. If the input level exceeds 0 dB, clipping occurs.(See "Monitoring recording and playback levels" on page 79.)

4 If you need to adjust the CD Audio input level, choose Options > Windows Recording Mixer to open the Windows Recording Control panel. Adjust the CD Audio input level as desired.

5 After you finish monitoring the input level, choose Options > Monitor Record Level.

To record from a CD internally:

1 In Edit View, create a new file.

2 Click the Record button .

3 Start the desired track in your CD player application.

4 When desired, stop recording in both Adobe Audition and the CD player application.

Setting the current-time indicator

The *current-time indicator* is a vertical, dotted line in the display window. You set the current-time indicator in order to start playback or recording at a specific point in a waveform.

When you work with multiple files in Edit View, you can use the Synchronize Cursor Across Windows command to retain the position of the current-time indicator between files. This command is useful if you switch between different versions of the same waveform during editing. In Multitrack View, you can use the Synchronize Clips With Edit View command to maintain the position of the current-time indicator when you switch between Multitrack View and Edit View.

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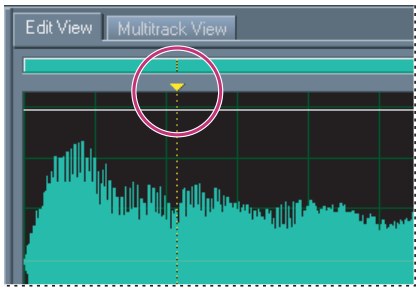

Current-time indicator

To set the current-time indicator:

Do one of the following in the display window:

- Click exactly where you want to set the current time.
- Position the pointer over the triangle above or below the current-time indicator. (This triangle is the current-time indicator's *handle*.) Drag the handle to the desired position in the timeline.

After you set the current-time indicator, you can save it as a cue for later reference. For more information, see "Working with cues" on page 96.

To synchronize the current-time indicator between waveforms:

In Edit View, choose Options > Synchronize Cursor Across Windows.

To synchronize the current-time indicator between Multitrack View and Edit View:

In Multitrack View, choose Options > Synchronize Clips With Edit View.

Monitoring time

Adobe Audition provides several features to help you monitor time during recording and playback. The *playback cursor*—a vertical, white line that appears in the display window—shows you the current time in the waveform. The Time window shows the current time in numerical format. The default display format is mm:ss:ddd (minutes:seconds:thousandths of a second), but you can easily change it. The display format is also used by the timeline along the bottom of the display window.

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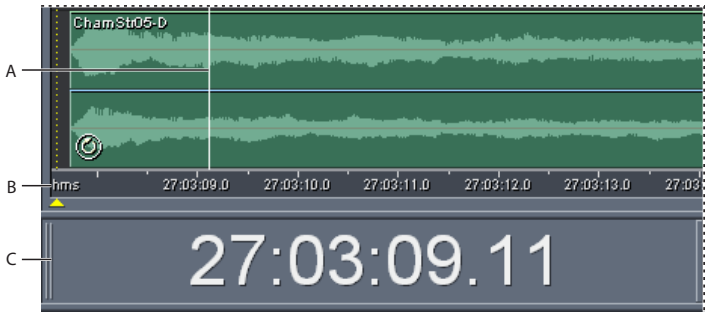

Features that help you monitor time
***A.** Playback cursor **B.** Timeline **C.** Time window*

To display the Time window:

Do one of the following:

- Choose Window > Time. A check mark indicates that the window is visible.
- Click the Hide/Show Time Window button in the View toolbar. (See "Using toolbars" on page 13.)

If you don't like the default location of the Time window, you can reposition it or detach it so it floats above the main window. (See "Using windows" on page 14.)

To change the time display format:

Choose View > Display Time Format, and choose the desired option:

- Decimal (mm:ss.ddd) displays time in minutes, seconds, and thousandths of a second.
- Compact Disc 75 fps displays time in the same format utilized by audio compact discs, where each second equals 75 frames.
- SMPTE 30 fps displays time in the SMPTE format, where each second equals 30 frames.
- SMPTE Drop (29.97 fps) displays time in the SMPTE drop-frame format, where each second equals 29.97 frames.
- SMPTE 29.97 fps displays time in the SMPTE non-drop-frame format, where each second equals 29.97 frames.
- SMPTE 25 fps (EBU) displays time using the standard European frame rate, where each second equals 25 frames.

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ug.book Page 71 Tuesday, March 16, 2004 1:29 PM

- SMPTE 24 fps (Film) displays time in a format where each second equals 24 frames, suitable for film.
- Samples displays time numerically, using as a reference the actual number of samples that have passed since the beginning of the edited file.
- Bars and Beats displays time in a musical measures format of bars:beats:ticks. To adjust the settings, choose Edit Tempo. For more information, see "Calculating the tempo of selected ranges" on page 199.
- Custom (X frames/sec) displays time in a custom format. To modify a custom format, choose Edit Custom Time Format, enter a number of frames per second for Custom Time Code Display, and click OK.

Using the transport controls

Just like many hardware-based audio recording and playback devices, Adobe Audition provides transport controls for playing, recording, stopping, pausing, fast forwarding, and rewinding waveforms and sessions.

Right-click the transport control buttons to set options for playing, recording, fast forwarding, and rewinding audio.

Transport controls

To show or hide the transport controls:

Do one of the following:

- Choose Window > Transport Controls. A check mark indicates that the controls are visible.
- Click the Hide/Show Transport Controls button in the View toolbar. (See "Using toolbars" on page 13.)

If you don't like the default location of the transport controls, you can reposition them or detach them so they float above the main window. (See "Using windows" on page 14.)

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ug.book Page 72 Tuesday, March 16, 2004 1:29 PM

Recording audio

You can record audio from a microphone or any signal you can plug into the Line In port of a sound card.

Note: *You may need to adjust the input signal to obtain the optimum recording and signal-to-noise levels. (See "Adjusting a sound card's levels" on page 82.)*

By default, Adobe Audition displays waveforms in real time while recording. However, if the recorded audio is choppy, deselect Live Update During Recording in the General tab of the Settings dialog box. (See "Setting Adobe Audition preferences" on page 43.)

Recording audio in Edit View

In Edit View, you can record audio into a new file or over existing audio. You can also disable the Record button so you don't start recording accidentally.

To record in Edit View:

1 Do one of the following:

- Create a new file. (See "Creating new audio files" on page 84.)
- In an existing file, place the current-time indicator where you want to start recording. (See "Setting the current-time indicator" on page 68.)

2 Click the Record button to begin recording.

3 Click the Stop button to stop recording.

To disable the Record button:

Right-click the Record button, and choose Disable Record Button. Repeat to reenable the button.

Using timed record mode

Use timed record mode to set start and stop times for recording. You can specify a maximum recording time and you can set a time for recording to start and stop automatically.

To enable or disable timed record mode:

Choose File > Timed Record Mode. Alternatively, right-click the Record button, and choose Timed Record Mode. A check mark indicates that timed record mode is enabled.

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ug.book Page 73 Tuesday, March 16, 2004 1:29 PM

To set start and stop times for recording:

1 Enable timed record mode.

2 Click the Record button .

3 Specify the maximum recording time:

- Select No Time Limit to record until you click the Stop button (or until disk space runs out).
- Select Recording Length to record for the duration you specify in the Recording Length box.

4 Specify when to start recording:

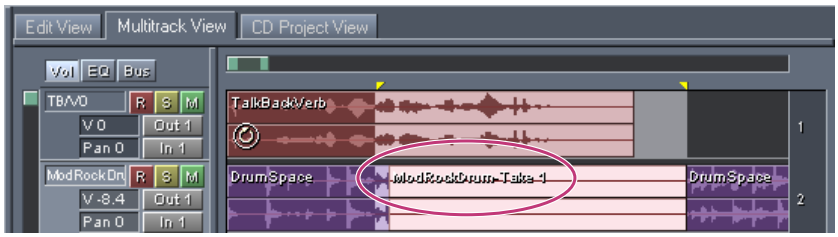
- Select Right Away to begin recording as soon as you click OK.
- Select Time/Date to begin recording at a time you specify (for example, to have Adobe Audition capture a radio broadcast at a certain time). Enter the starting time and date in the appropriate text boxes, and set the desired time and date options.

5 Click OK.

Recording audio in Multitrack View

In Multitrack View, you can record audio on multiple tracks by overdubbing. When you overdub tracks, you can hear previously recorded tracks and play along with them to create sophisticated, layered compositions.

Each recording becomes a new audio clip on a track. If you are unsatisfied with a section of a recorded clip, you can select that section and punch in a new recording—leaving the remainder of the original clip intact. For particularly important or difficult sections, you can punch in multiple takes (different versions), and then select the take with the best performance.



A take created with the Punch In command

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To record a new clip in a track:

1 In the track controls area for the track, click the In 1 button, select the desired input of your sound card, and then click OK.

2 Click the Record-enable button for the track.

3 To simultaneously record on multiple tracks, repeat steps 1-2 for each track.

4 Position the current-time indicator at the desired starting point for recording, or select the range where you want to record the clip.

5 Click the Record button to begin recording.

6 Click the Stop button ■ to stop recording.

To record in a loop:

1 Specify the input source, track, and starting point (or range) for recording, as described in the previous procedure.

2 Right-click the Record button, and choose one of the following options:

- Loop While Recording (View or Sel) to loop when the cursor reaches the end of the viewable range of track. If a range is selected, looping occurs when the cursor reaches the end of the range.
- Loop While Recording (Entire or Sel) to loop when the cursor reaches the end of the track. If a range is selected, looping occurs when the cursor reaches the end of the range.

3 Click the Record button to begin recording.

4 Click the Stop button ■ to stop recording.

If you use either of the Loop While Recording options for punching in audio, a new take is created with each loop.

To punch into a range of a clip:

1 In the track display, select the range of the clip.

2 Choose Edit > Punch In.

3 Position the current-time indicator a few seconds prior to the selected range.

4 In the Transport Controls window, click the Record button.

5 To punch in multiple takes, repeat step 4 for each take.

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ug.book Page 75 Tuesday, March 16, 2004 1:29 PM

Note: You can't punch into a loop-enabled clip. For information about disabling loops, see "Setting impermanent loop properties in Multitrack View" on page 202.

To select from multiple takes in a clip:

1 Select the clip.

2 Choose Edit > Take History, and then select the desired take.

To merge a selected take into a clip:

Choose Edit > Take History > Merge This Take (Destructive).

Note: Merging destructively adds a 30 millisecond crossfade at take edges.

To delete a selected take:

Choose Edit > Take History > Delete This Take.

Playing audio

Adobe Audition provides several ways to play audio, including using the transport controls to play the currently active file, using the Organizer window to preview files, and using the Windows Run command to start Adobe Audition and begin playing a file.

Playing audio by using the transport controls

The transport controls provide several options for playing the currently active file. For example, you can play just the visible section of a waveform, the duration from the current-time indicator to the end of the file, or the entire waveform. In addition, you can set preroll and postroll options to play a selection with just a bit of audio preceding or following it.

To start playback without using the transport controls, press the space bar. Press the space bar again to stop playback.

To play a range of audio:

Select the range you want to play, and click the Play button ▶ in the Transport Controls window.

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ug.book Page 76 Tuesday, March 16, 2004 1:29 PM

To play from the current-time indicator to the end of the current view:

Set the current-time indicator where you want playback to start, and click the Play button in the Transport Controls window.

To play from the current-time indicator to the end of the file:

Set the current-time indicator where you want playback to start, and click the Play To End button in the Transport Controls window.

To play the visible portion of the file:

Right-click the Play button or Play To End button, and choose Play View. Then click the button again to start playback.

To play an entire file:

Right-click the Play button or Play To End button, and choose Play Entire File. Then click the button again to start playback.

To loop audio during playback:

Do one of the following:

- To play the currently-visible portion of the audio in a continuous loop, click the Play Looped button in the Transport Controls window.
- To loop the entire waveform or session (or just the selected range), right-click the Play Looped button and choose Play Entire (or Selection). Then click the button again to start playback.

Note: *By default, the display window scrolls in sync with playback that extends beyond the visible section of a waveform. In the General tab of the Settings dialog box, you can set options for auto-scrolling or you can disable this feature. (See "Setting Adobe Audition preferences" on page 43.)*

Using preroll and postroll during playback (Edit View only)

In Edit View, you can play back the audio just before and after a selected range. This audio is known as *preroll* and *postroll*. Playing preroll and postroll is useful for fine-tuning selections and listening to transitions without destroying a selection. By default, the duration of preroll and postroll is one second; however, you can adjust this duration to best meet your needs.

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To play a selected range of audio with preroll and postroll:

1 In Edit View, right-click the Play button or the Play To End button in the Transport Controls window, and choose one of the following options: Play Preroll and Postroll, Play Preroll and Selection, Play Postroll, or Play Preroll, Postroll, and Selection

2 Click the button again to start playback.

You can also use keyboard shortcuts to play preroll and postroll. For information on specific keyboard shortcuts, "Keys for playing audio" on page 263.

To set a duration for preroll and postroll:

1 In Edit View, right-click the Play button or the Play To End button in the Transport Controls window.

2 Choose Preroll and Postroll Options.

3 In the Edit View–Play section of the Preroll and Postroll Options dialog box, specify a duration for preroll and postroll.

4 Click OK.

Previewing audio by using the Organizer window

The Files tab in the Organizer window provides several play options that make it easy to preview loops and other files. These options are particularly handy when you work in Multitrack View because they let you preview loops at the session tempo. For more information on using the Files tab in the Organizer window, see "Organizing files" on page 24.

To preview a file:

1 Make sure that the advanced options—including the preview and sorting controls—appear in the Files tab of the Organizer window. If they don't, click the Advanced Options button at the top of the Files tab.

2 Select the file you want to preview, and then click the Play button. Click the Stop button to stop the preview. Use the volume slider to adjust the volume of the preview.

To enable auto-play:

Click the Auto-play button on the Files tab. Adobe Audition automatically previews files you select. To disable auto-play, click the Auto-play button again.

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ug.book Page 78 Tuesday, March 16, 2004 1:29 PM

To preview a file at the session tempo (Multitrack View only):

In Multitrack View, select Follow Session on the Files tab, and then click the Play button or enable auto-play, and select a file.

Note: Only files that are loop-enabled can be previewed at the session tempo. Loop-enabled files are identified with a loop icon in the Files tab.

To enable continuous loop preview:

Click the Loop button ∞ on the Files tab, and then click the Play button or enable auto-play, and select a file. To disable continuous loop preview, click the Loop button again.

Playing audio by using the Windows Run command

You can start Adobe Audition and begin playing a file by using the Windows Run command. Before using the command, make sure that Auto-Play On Command-Line Load in the General tab of the Settings dialog box is selected. (See "Setting Adobe Audition preferences" on page 43.)

To play audio from the command line:

In the Windows Run dialog box, type the following text, and click OK:

"*[drive]*:\Program Files\Adobe\Audition 1.5\Audition.exe" "*[path to file]*"

For example, type the following to play the TalkBackVerb loop file:

"c:\Program Files\Adobe\Audition 1.5\Audition.exe" "c:\Program Files\Adobe\Audition 1.5\Audition Theme\TalkBackVerb.cel"

Stopping, pausing, and adjusting the playback cursor

The transport controls provide buttons for stop recording and playback, pausing recording and playback, and adjusting the playback cursor.

To stop playback without using the transport controls, press the spacebar. Press the spacebar again to start playback.

To stop playing or recording audio:

Click the Stop button ■ in the Transport Controls window.

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ug.book Page 79 Tuesday, March 16, 2004 1:29 PM

To pause playing or recording audio:

Click the Pause button in the Transport Controls window. Click the Pause button again to resume playback or recording.

To adjust the playback cursor:

Click one of the following buttons in the Transport Controls window:

- The Go to Beginning button places the playback cursor at the beginning of the waveform or session.
- The Rewind button shuttles the playback cursor backward in time. This function supports scrubbing, meaning that on some sound cards, the audio file plays back at a lower volume as the playback cursor shuttles over the waveform or session.

Right-click the Rewind button to set the rate at which the cursor moves.

- The Fast Forward button shuttles the playback cursor forward in time. This function supports scrubbing, meaning that on some sound cards, the audio file plays back at a lower volume as the playback cursor shuttles over the waveform or session.

Right-click the Fast Forward button to set the rate at which the cursor moves.

- The Go to End button places the playback cursor at the end of a waveform (in Edit View) or at the end of the list clip in a session (in Multitrack View).

Monitoring recording and playback levels

Adobe Audition provides the Level Meters to help you monitor the amplitude of the signal during recording and playback. If the amplitude is too high, *clipping* occurs and results in distortion; if the amplitude is too low, the sound quality is reduced.

If you find that the signal is too high or low during recording and playback, you can adjust the input and output levels of your sound card.

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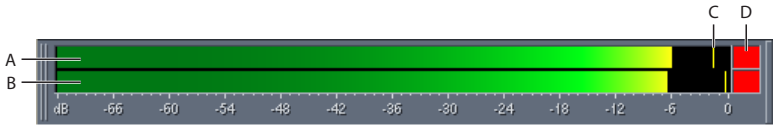
ug.book Page 80 Tuesday, March 16, 2004 1:29 PM

Using the Level Meters

The Level Meters represent the incoming signal in *dBFS* (decibels below full scale), where a level of 0 dB is the maximum amplitude possible before clipping occurs. Yellow peak indicators remain for 1.5 seconds to allow for reading of the peak amplitude. If clipping does occur, the clip indicator to the right of the meter lights up and stays on until you clear it. When stereo audio is displayed, the top meter represents the left channel, and the bottom represents the right.

You can customize the Level Meters in a variety of ways, such as changing the decibel range, showing valley (minimum amplitude) indicators, and changing the mode of the peak indicators.

Right-click the Level Meters to set metering options.



The Level Meters
A. Left channel B. Right channel C. Peak indicators D. Clip indicators

To show or hide the Level Meters:

Do one of the following:

- Choose Window > Level Meters. A check mark indicates that the Level Meters are visible.
- Click the Hide/Show Level Meters button in the View toolbar. (See "Using toolbars" on page 13.)

If you don't like the default location of the Level Meters, you can reposition them or detach them so they float above the main window. (See "Using windows" on page 14.)

To start or stop monitoring the levels of an input source:

Choose Options > Monitor Record Level, or double-click the Level Meters.

To disable or enable the Level Meters during recording or playback:

Choose Options > Show Levels On Play And Record. Disabling the Level Meters improves performance on lower specification computers.

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ug.book Page 81 Tuesday, March 16, 2004 1:29 PM

To clear a clip indicator:

Click the clip indicator, or right-click the Level Meters and choose Clear Clip Indicator.

Note: *The clip indicators always light up if clipping occurs, but if Adjust For DC is enabled, the indicators light up if audio has a DC offset.*

To adjust for DC offset:

Right-click the Level Meters, and choose Adjust For DC.

Many sound cards record audio with a slight DC offset, meaning that the center of the waveform being recorded is a little above or below the center of the waveform display. This offset can dramatically throw off the level meters since the offset amount could be interpreted as a constant sound at that volume. You should have this option enabled when recording.

To show or hide valley indicators:

Right-click the Level Meters, and choose Show Valleys.

If the valley indicators are close to the peak indicators, the dynamic range (the difference between the quietest and loudest sounds) is low. If they're spread far apart, the dynamic range is high.

To change the decibel range of the Level Meters:

Right-click the Level Meters, and choose a Range option.

To change the mode of peak indicators:

Right-click the Level Meters, and choose one of the following options:

- Dynamic Peaks causes the yellow peak level indicators to reset to a new peak level after 1.5 seconds, letting you easily see the peak amplitude "right now." As the audio gets quieter, the peak indicators start backing off.
- Static Peaks keeps the peak levels from being reset, letting you retain the maximum amplitude of the signal since monitoring, playing, or recording began. The peak can still be reset manually at any time by clearing the clip indicators (that is, by clicking the clip indicator at the right).

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ug.book Page 82 Tuesday, March 16, 2004 1:29 PM

Select Static Peaks as a great way to find out how loud a song will get before you record it. Just start monitoring levels and then play the song. After the song ends, the peak indicators show the volume of the loudest part.

Adjusting a sound card's levels

Adobe Audition doesn't directly control a sound card's record levels (input gain) and playback levels (output volume). Instead, you can adjust these levels with the mixer application that comes with the sound card or with the mixer built into Windows. You may need to adjust levels if recordings are too quiet (causing unwanted noise), too loud (leading to clipped, distorted sound), or not audible when played in Adobe Audition.

To get the best sounding results, you should record audio as loud as possible without clipping. Try to keep the loudest point somewhere between –2 dB and 0 dB when setting the recording levels.

To adjust a sound card's record and playback levels by using Windows:

1 Open the Windows Volume Control program.

You can usually access this program in the Programs > Accessories > Entertainment (or Multimedia) menu of the Windows Start menu. On many systems, you can also double-click the speaker icon in the system tray to access the Volume Control program, which resembles a small mixing board with vertical sliders.

2 To adjust the sound card's playback (output) level, turn up the sliders on the Windows mixer to the desired volume. Make sure that Mute underneath both sliders isn't selected.

3 To adjust the sound card's record (input) level, choose Options > Properties in Volume Control. Select Recording and click OK. Be sure that the input source you want to use is selected, and adjust other sliders on the Windows mixer as needed.

To quickly access the Record section of the Windows mixer, choose Options > Windows Recording Mixer in Adobe Audition.

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ug.book Page 83 Tuesday, March 16, 2004 1:29 PM

Chapter 4: Editing Audio

Adobe Audition provides a powerful and easy-to-use interface for preforming a variety of editing tasks, such as copying, pasting, and deleting; adding and removing silence; generating noise and tones; changing the sample type; and adding information to audio files.

About editing audio

When you open an audio file in Edit View, you see the *waveform display*, a visual representation of the sound wave, or *waveform*. If you open a stereo file, the waveform for the left channel appears at the top and the waveform for the right channel appears at the bottom. If you open a mono file, the waveform utilizes the total height of the waveform display. The peaks and valleys in the waveform represent positive and negative air pressure. Quiet audio has both lower peaks and lower valleys than loud audio.

For background information on working with digital audio, see "Sound fundamentals" on page 267.

Stereo waveform in Edit View

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ug.book Page 84 Tuesday, March 16, 2004 1:29 PM

Many editing tasks require that you select a precise range of a waveform. When selecting a range, you'll probably want to zoom in to view the waveform in more detail. (See "Zooming" on page 17.) Adobe Audition provides a variety of ways to select audio data precisely, such as adjusting selections to zero-crossings, finding beats, and using snapping. (See "Selecting audio data" on page 87.)

As you edit a waveform, keep in mind that you can undo your changes until you save the file. (See "Undoing and redoing changes" on page 22.)

Creating new audio files

The File > New command lets you create an empty audio file. Doing so is useful when you want to paste audio into an empty file before you edit it or when you want to record audio into a new file.

You can quickly create a new file from a selection by choosing Edit > Copy to New. (See "Copying audio data" on page 92.)

To create a new audio file:

1 Choose File > New. Alternatively, click the New File button in the toolbar.

2 Select a sample rate in the list, or type a custom sample rate in the text box.

The sample rate determines how many frequencies can be encoded in the audio signal. (Higher sampling rates mean a wider bandwidth.) For more information, see "About sample rates" on page 110.

3 Select a number of channels:

- Mono creates a waveform with just one channel of audio information. This setting is good for voice-only recordings.
- Stereo creates a waveform with separate right and left channels. This setting is usually best for music recordings. Because stereo waveforms contain twice as much data as mono waveforms, they consume twice as much storage space.

4 Select a resolution, and click OK:

- 8-bit creates a waveform where quality is not much of a concern, but small file size is. 8-bit waveforms are usually fine for telephony applications or for embedded sounds in Web pages. Although they tend to be noisier than their 16-bit counterparts, they're half the size.

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ug.book Page 85 Tuesday, March 16, 2004 1:29 PM

- 16-bit produces a CD-quality waveform. This setting is suitable for most broadcast and music recordings.
- 32-bit creates a waveform that supports the most precise audio processing, and 32-bit is the recommended resolution for editing files in Adobe Audition. After you edit a file, you can downsample it to 16- or 8-bit for output and achieve better results than if you edit an 8- or a 16-bit file. (See "Changing the bit depth" on page 113.)

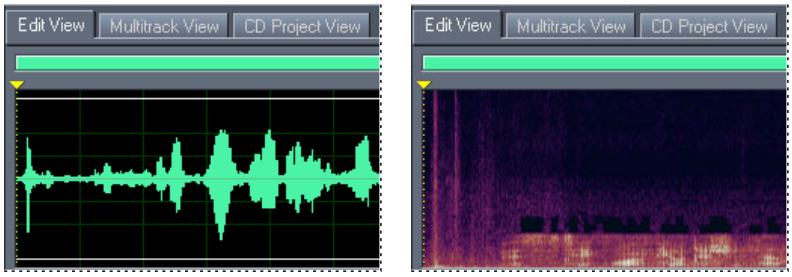
Note: *Older sound cards might not be able to play 32-bit files properly. To check the capabilities of your sound card, choose Options > Device Properties. If your sound card doesn't support 32-bit files, you can limit playback to 16-bits while retaining the 32-bit depth internally. (See "Setting properties for audio output devices" on page 37.)*

Viewing waveforms

The waveform display in Edit View shows you a visual representation of a waveform. By default, this representation shows the amplitude of a waveform over time. However, you can view the frequency of a waveform over time by switching to Spectral View. You can also control the scale with which Adobe Audition measures the amplitude or frequency of waveforms.

Switching between Waveform View and Spectral View

The waveform display offers two ways to represent audio data: Waveform View and Spectral View.



Waveform View and Spectral View

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ug.book Page 86 Tuesday, March 16, 2004 1:29 PM

- Waveform View (the default) displays a waveform as a series of positive and negative peaks. The *x*-axis (horizontal ruler) represents time, and the *y*-axis (vertical ruler) measures spikes, or increased amplitude, in a waveform."
- Spectral View displays a waveform by its frequency components, where the *x*-axis represents time and the *y*-axis (vertical ruler) measures frequency. This view lets you analyze audio data to see which frequencies are most prevalent. The greater a signal's amplitude component within a specific frequency range, the brighter the displayed color. Colors range from dark blue (meaning that the frequencies are very low in amplitude) to bright yellow (meaning that the frequencies are high in amplitude).

To switch between Waveform View and Spectral View:

Choose View > Waveform View or View > Spectral View. Alternatively, click the Toggle Between Waveform And Spectral Views button in the toolbar.

Adobe Audition lets you customize certain features of Waveform View and Spectral View. For example, you can show or hide grid lines in Waveform View and change the resolution in Spectral View. For more information, see "Display options" on page 50.

Changing the vertical scale

In Waveform View, the vertical ruler shows the decibel value of the audio data. However, you can change the scale of the ruler to Sample Values, Normalized Values, or Percentage.

Note: *In Spectral View, the vertical scale is always in hertz (Hz).*

To change the scale of the vertical ruler:

Choose View > Vertical Scale Format, and choose the desired scale:

- Sample Values indicates amplitude as the data's exact sample value of the data.
- Normalized Values indicates amplitude on a normalized scale value that ranges from – 1 to 1.
- Percentage indicates amplitude on a percentage scale value that ranges from –100% to 100%.
- Decibels indicates amplitude using a decibel scale value that ranges from –Infinity to Zero.

Double-click the vertical ruler to cycle through the scales.

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ug.book Page 87 Tuesday, March 16, 2004 1:29 PM

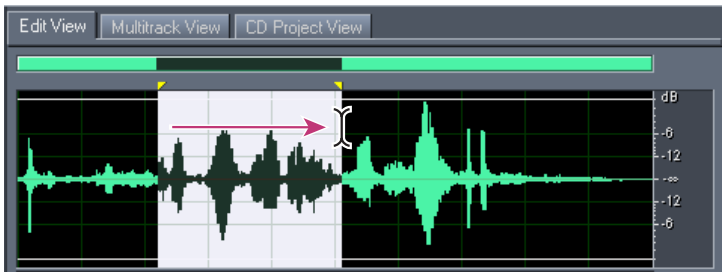
Selecting audio data

To edit a waveform, you must first select the audio data that you want to modify. Adobe Audition provides several methods for making and adjusting selections.

Using cues ranges can save you time when making selections. (See "Working with cues" on page 96.)

Selecting with the mouse

You can select a range of audio data by dragging in the waveform display. When precision is important, you may want to zoom in to view the waveform in more detail. (See "Zooming" on page 17.)



Dragging to select a range

To select a range of a waveform:

Drag to select the desired range of the waveform.

To extend or shorten a selection:

Shift-click the end of the selection that you wish to modify, and drag to extend or shorten it.

Note: *If you prefer, you can right-click to extend or shorten a selection. To enable this feature, select Extend Selection in the General tab of the Settings dialog box. (See "Setting Adobe Audition preferences" on page 43.)*

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ug.book Page 88 Tuesday, March 16, 2004 1:29 PM

To select a range in only one channel:

Do one of the following:

- Drag near the top of the left (upper) channel. The cursor displays an *L* icon to indicate the left channel.
- Drag near the bottom of the right (lower) channel. The cursor displays an *R* icon to indicate the right channel.

To select the visible range of a waveform:

Double-click in the waveform display.

Selecting all of a waveform

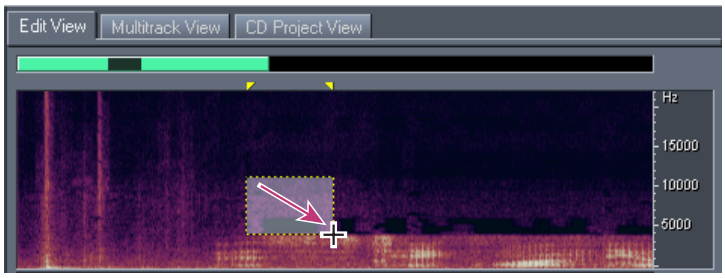
The Select Entire Wave command lets you select all of the audio data in a waveform. You can use the mouse to do this as well.

To select all of a waveform:

Choose Edit > Select Entire Wave, or triple-click in the waveform display.

Selecting audio frequencies in Spectral View

When working in Spectral View, you can use the Marquee Selection tool to select audio data within specific frequencies. This method allows for band-limited editing and processing, as well as greater flexibility in audio restoration work. For example, if you detect an audio anomaly or error, you can select and edit just the affected frequencies, resulting in superior results and faster processing.



Making a marquee selection in Spectral View

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ug.book Page 89 Tuesday, March 16, 2004 1:29 PM

To make a marquee selection:

1 In Spectral View, click the Marquee Selection button ⬚ in the toolbar.

If this button isn't visible, choose View > Toolbars > Spectral Selection.

2 Drag in the waveform display to select the desired audio data.

When making a marquee selection in a stereo waveform, the selection is applied to both channels. To select audio data in just one channel, choose Edit > Edit Channel, and then choose Edit Left Channel or Edit Right Channel.

To move a marquee selection:

Position the cursor in the selection, and drag it to the desired location.

To resize a marquee selection:

Position the cursor on the corner or edge of the selection, and drag it to the desired size.

Adjusting selections to zero-crossing points

For many editing tasks, such as deleting or inserting audio in the middle of a waveform, the best places to make selections are the points where the amplitude is zero (called *zero-crossings*). Selecting the zero-crossing points reduces the chance that an edit will create an audible pop or click. You can easily adjust a selection to the closest zero-crossing points by using a Zero Crossing command.

To adjust a selection to zero-crossing points:

Choose Edit > Zero Crossing, and choose one of the following commands:

- Adjust Selection Inward adjusts both range boundaries inward to the next zero-crossing point. Alternatively, click the Zero Crossing button in the toolbar.
- Adjust Selection Outward adjusts both range boundaries outward to the next zero-crossing point.
- Adjust Left Side To Left adjusts the left range boundary leftward to the next zero-crossing point.
- Adjust Left Side To Right adjusts the left range boundary rightward to the next zero crossing point.

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ug.book Page 90 Tuesday, March 16, 2004 1:29 PM

- Adjust Right Side To Left adjusts the right range boundary leftward to the next zero crossing point.
- Adjust Right Side to Right adjusts the right range boundary rightward to the next zero crossing point.

Finding beats

For some editing tasks, such as constructing drum loops and similar musical phrases, you need to select audio between beats. You can usually pick out where the beats are by looking for the peaks in a waveform. You can also use a Find Beats command.

Once you find beats, you can save them as Beat Cues, making it easy to locate the beats again. (See "Working with cues" on page 96.)

To find the beginning of a beat:

1 Click in the waveform display to the left of the first beat you want to find.

2 Choose Edit > Find Beats > Find Next Beat (Left). The cursor moves to the beginning of the next beat.

To select audio between beats:

1 Find the beginning of a beat.

2 Choose Edit > Find Beats > Find Next Beat (Right) to select from the current cursor position to the next beat.

3 If you want to select more than one beat, choose Edit > Find Beats > Find Next Beat (Right) again. Each time you choose this command, Adobe Audition adds the next beat to the selection.

When you select audio to construct a loop, click the Play Looped button in the transport controls to preview the loop. After any necessary tweaking, you can then save, paste, or add the loop to the Cue List.

To adjust the settings that Adobe Audition uses to find beats:

Choose Edit > Find Beats > Beat Settings. Enter new values for Decibel Rise and Rise Time, and click OK.

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ug.book Page 91 Tuesday, March 16, 2004 1:29 PM

For finding beats with material that has fast transient attacks, such as drums, specify a quick Rise Time and a high Decibel Rise so as not to cut off the beginning of the attack. For material with softer attacks, such as bass, the Rise Time can be slightly slower relative to Decibel Rise.

Snapping

Snapping causes selection boundaries, as well as the current-time indicator, to move to items such as cues, ruler ticks, zero-crossing points, and frames. Enabling snapping helps you make accurate selections; however, if you prefer, you can disable snapping for specific items.

To enable or disable snapping:

Choose Edit > Snapping, and choose any of the following commands. A check mark indicates that a command is enabled:

- Snap To Cues allows the cursor to snap to a cue point. For more information on defining cues, see "Working with cues" on page 96.
- Snap To Ruler (Coarse) allows the cursor to snap only to the major numeric divisions (decimal, SMPTE, samples, and so on) in the timeline.

Note: *You can enable only one Snap To Ruler command at a time.*

- Snap To Ruler (Fine) allows the cursor to snap to each of the subdivisions (decimal, SMPTE, samples, and so on) within the timeline. Zooming in (by right-clicking as you drag across the timeline) breaks the display down into more accurate subdivisions, letting you place the cursor more accurately within the timeline.
- Snap To Zero Crossings allows the cursor to snap to the nearest place where the waveform crosses the center line (in other words, the zero amplitude point).
- Snap To Frames (Always) allows the cursor to snap to a frame boundary, as long as the time format is measured in frames (such as Compact Disc and SMPTE). This command is especially handy for working on audio for CD.

You can access snapping commands by right-clicking the timeline.

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ug.book Page 92 Tuesday, March 16, 2004 1:29 PM

Specifying which channel of a stereo waveform to edit

By default, Adobe Audition applies selections and edits to both channels of a stereo waveform. However, you can easily select and edit just the left or right channel of a stereo waveform.

To specify which channel you want to edit:

Do either of the following:

- Choose Edit > Edit Channel, and choose which channel you want to edit.
- Click the Edit Left Channel button, the Edit Right Channel button, or the Edit Both Channels button in the View toolbar. (See "Using toolbars" on page 13.)

Copying, cutting, pasting, and deleting

Adobe Audition supports all of the basic editing functions, as well as several commands designed specifically for audio editing.

Choosing a clipboard

Adobe Audition gives you access to five internal clipboards for temporary data storage. Each works similarly to the Windows clipboard, except that they can handle more data at a faster rate.

To choose a clipboard:

Choose Edit > Set Current Clipboard, and choose a clipboard.

Choose the Windows clipboard if you want to copy audio data to other Windows applications.

Copying audio data

The Copy command lets you copy audio data to the active clipboard. The Copy To New command lets you copy and paste the data to a new file in one step.

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ug.book Page 93 Tuesday, March 16, 2004 1:29 PM

To copy audio data:

1 In the waveform display, select the audio data you want to copy. Or, to copy the entire waveform, deselect all audio data.

2 Choose Edit > Copy or Edit > Copy To New. Alternatively, click the Copy button in the toolbar.

Cutting audio data

The Cut command lets you remove audio data from the current waveform and copy it to the active clipboard.

To cut audio data:

1 Select the audio data you want to cut. Or, to cut the entire waveform, deselect all audio data.

2 Choose Edit > Cut. Alternatively, click the Cut button in the toolbar.

Pasting audio data

The Paste command lets you paste audio data from the active clipboard to the current waveform. If the format of the data on the clipboard differs from the format of the file it's being pasted into, Adobe Audition automatically converts the format before pasting the data.

The Paste To New command lets you create a new file and insert audio data from the active clipboard. The new file automatically inherits the properties (sample rate, sample frequency, and so on) from the original clipboard material.

The Highlight After Paste option in the General tab of the Settings dialog box determines whether or not data is highlighted after you paste it into a file.

To paste audio data into the current file:

1 In the waveform display, place the cursor where you want to insert the audio data or select the audio data you want to replace.

2 Choose Edit > Paste. Alternatively, click the Paste button in the toolbar.

To paste audio data into a new file:

Choose Edit > Paste To New.

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ug.book Page 94 Tuesday, March 16, 2004 1:29 PM

Mixing audio data when pasting

The Mix Paste command lets you mix audio data from the clipboard or a file with the current waveform. If the format of the data on the clipboard differs from the format of the file it's being pasted into, Adobe Audition automatically converts the format before pasting the data.

The Mix Paste command provides a quick alternative to using the more powerful and flexible multitrack functions in Adobe Audition.

To mix audio data with the current waveform:

1 In the waveform display, place the cursor where you want to start mixing the audio data. Alternately, select the audio data you want to replace.

2 Choose Edit > Mix Paste. Alternatively, click the Mix Paste button in the toolbar.

3 Set the following options as desired, and click OK.

Volume Adjusts the sound level of the left and right channels before pasting. Move the volume sliders, or enter a percentage in the text boxes to the right of them.

Paste in single channels (either left or right) by adjusting the level of the opposite channel to zero.

Invert Turns that channel of the waveform upside-down. (Any samples above the center line are placed below it, and those below the center line are placed above it.)

This option is handy when you want to take the difference between two samples (or subtract one signal from another).

Lock Left/Right Locks the volume sliders so that they move together.

Insert Inserts audio at the current location or selection, replacing any selected data. If no data is selected, Adobe Audition inserts audio at the cursor location, moving any existing data to the end of the inserted material.

Overlap Mixes audio at the selected volume level with the current waveform. If the audio is longer than the current waveform, the current waveform is lengthened to accommodate the pasted audio.

Replace Overdubs the audio beginning at the cursor location, and replaces the existing material thereafter for the duration of audio. For example, pasting 5 seconds of material replaces the first 5 seconds after the cursor.

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ug.book Page 95 Tuesday, March 16, 2004 1:29 PM

Modulate Modulates the audio with the current waveform for an interesting effect. The result is similar to overlapping, except that the values of the two waveforms are multiplied by each other, sample by sample, instead of added.

You can create fantastic combo effects by selecting part of a wave and using the Mix Paste command with Modulate selected. The selection is modulated with the audio signal on the clipboard.

Crossfade Applies a fade to the beginning and end of the pasted audio. Enter a value to specify how many milliseconds of the audio are faded.

Use this option for smoother transitions to and from pasted audio.

From Clipboard *[number]* Pastes audio data from the active internal clipboard.

From Windows Clipboard Pastes audio data from the Windows clipboard. If the Windows clipboard contains no audio data, this option is disabled.

From File Pastes audio data from a file. Click Select File to browse for the file.

Loop Paste Pastes audio data the specified number of times. If the audio is longer than the current selection, the current selection is automatically lengthened accordingly.

Deleting audio data

Adobe Audition provides two methods for deleting audio: The Delete Selection command lets you remove a range from a waveform, whereas the Trim command lets you remove unwanted audio from both sides of the selected audio.

Note: *Deleted data doesn't go to the clipboard and can be retrieved only by choosing Edit > Undo or File > Revert To Saved, but only if you haven't saved the file since deleting the data.*

To delete audio data:

1. In the waveform display, select the audio data you want to delete.
2. Choose Edit > Delete Selection. Alternatively, click the Delete button in the toolbar.

To trim audio data:

1. In the waveform display, select the audio data you want to keep.
2. Choose Edit > Trim. Alternatively, click the Trim button in the toolbar.

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ug.book Page 96 Tuesday, March 16, 2004 1:29 PM

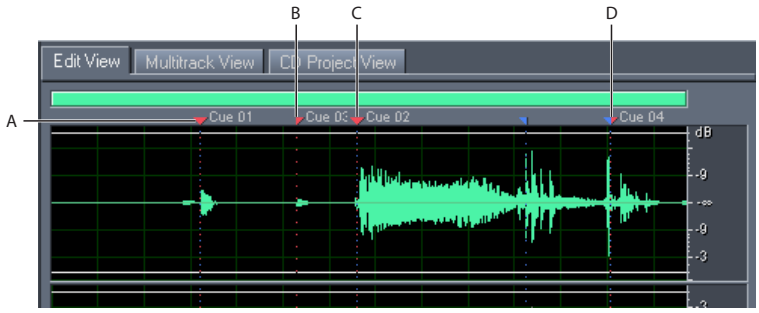
Working with cues

Cues are locations in a waveform that you define. Using cues makes it easy to navigate within a waveform in order to make a selection, perform edits, or play back audio.

About cues

In Adobe Audition, a cue can be either a *point* or a *range*. A point refers to an exact position within a waveform (for instance, 1:08.566 from the start of the wave). A range has both a start time and an end time (for example, all of the waveform from 1:08.566 to 3:07.379). If a cue is a range, you can drag its beginning and end points to different times.

Cues have triangular handles that appear at the top and bottom of the waveform display. You use cue handles to select and adjust cues. You can also right-click a cue handle to view commands for working with cues.



Examples of cues
***A.** Cue handle **B.** Cue point **C.** Cue range **D.** Nonsplit cue range*

***Note:** To preserve cues when you save a file, make sure that you select Save Extra Non-Audio Information.*

Defining and selecting cues

You use the Cue List window to define and select cues. You can also define cues by using context commands and keyboard shortcuts.

To display the Cue List:

Choose Window > Cue List. Alternatively, click the Hide/Show Cue List button in the toolbar.

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ug.book Page 97 Tuesday, March 16, 2004 1:29 PM

To define a cue:

1 Do one of the following:

- Place the cursor exactly where you want the cue point to be in the waveform display.
- Select the audio data you want to define as a cue range in the waveform display.

2 Click Add in the Cue List window. Alternatively, click the Add To Cue List button in the toolbar.

To select cues:

Do one of the following:

- Click a cue in the cue list.
- Double-click a cue handle in the waveform display.
- To select adjacent (contiguous) cues, click the first cue you want to select in the cue list, and then Shift-click the last.
- To select nonadjacent (noncontiguous) cues, Ctrl-click them in the cue list.

Playing cues

The Auto Play feature in the Cue List window causes Adobe Audition to automatically play cues when you select them.

To enable or disable the Auto Play feature for cues:

Click the Auto Play button in the Cue List window.

Choosing a cue type

Adobe Audition provides four cue types. All four can be ranges as well as points, although it doesn't really make sense for index cues to be ranges. Consider the following when choosing a cue type:

- Basic Cues lets you mark important sections of a waveform for later reference (for example, to remind yourself of an editing point). Basic cues are also useful for specifying stop and start positions for the play list.

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ug.book Page 98 Tuesday, March 16, 2004 1:29 PM

- Beat Cues is just like Basic Cues, but you use it to mark musical beats. Beat cues are a very powerful feature because an audio file saved with them allows the beat mapping loop method to be very accurate. For more information on creating and using loops, see "About loops" on page 197.
- Track Cues lets you indicate a split in tracks for an audio compact disc. Use these cues only for burning CDs. (See "Inserting tracks" on page 258.)
- Index Cues lets you set markers within a CD track. (Some CD players offer controls for cueing indexes.) Also, the time between the track cue that begins a track and the first index cue in that track shows up on the player as "negative time."

To change the cue type:

1 Select a cue.

2 Click Edit Cue Info in the Cue List window.

3 Choose a cue type from the Type menu.

Alternatively, right-click the cue handle, and choose a cue type from the context menu.

Naming cues

After you create a cue, you can rename it and add descriptive information.

To rename a cue and add a description:

1 Select a cue.

2 Click Edit Cue Info in the Cue List window.

3 Do either or both of the following:

- Enter a new name in the Label text box.
- Enter a description in the Desc text box.

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ug.book Page 99 Tuesday, March 16, 2004 1:29 PM

Adjusting cues

You can easily adjust the position of cues, as well as the duration of range cues.

To reposition a cue:

Do one of the following:

- For point cues, drag the cue handle to a new location in the waveform display.
- For range cues, drag the red start handle to a new location in the waveform display.
- Select the cue, and click Edit Cue Info in the Cue List window. Enter a new value in the Begin text box.

To change the duration of a range cue:

- Drag the blue end handle to a new location in the waveform display.
- Select the cue, and click Edit Cue Info in the Cue List window. Enter a new value in the End or Length text box.

Merging, converting, and deleting cues

Adobe Audition lets you merge cues, and it also lets you convert point cues to range cues, and vice versa. If you find that you don't need a cue, you can delete it.

To merge cues:

1 Select the cues you want to merge. You can select only two cue ranges to merge, but you can select any number of cue points.

2 Click Merge in the Cue List window.

Note: *The new merged cue inherits its name from the first cue. You lose any information in the Label and Desc text boxes for the subsequent merged cue.*

To convert a point cue to a range cue:

Right-click the cue handle, and choose Make Range. The cue handle splits into two handles.

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ug.book Page 100 Tuesday, March 16, 2004 1:29 PM

To convert a range cue to a point cue:

Right-click a cue handle, and choose Make Point. The two parts of the range cue handle merge into a single handle, with the start time of the range becoming the time for the point cue.

To delete cues:

1 Select one or more cues.

2 Click Del in the Cue List window. Alternatively, right-click the cue handle, and choose Delete.

Batch processing cues

You can use the Batch feature in the Cue List window to add silence between cues and save the audio between cues to new files.

To batch process cues:

1 Select one or more cues. At least one of the cues you select must be a range.

2 Click Batch in the Cue List window.

3 Set the following options as desired, and click OK:

Set Amount Of Silence Adds silence between cue points in the waveform. Enter the desired values (measured in seconds) in the Add Silence Before and Add Silence After text boxes.

Save To Files Saves the audio between cue points to new files.

Filename Prefix Specifies a prefix for the new files. Adobe Audition automatically adds numbers after the prefix (phrase02, phrase03, and so on) as well as the correct extension based upon the output format you choose.

Destination Folder Specifies the folder where Adobe Audition places the new files. Click Browse to specify a different folder.

Output Format Specifies the desired output format for the new files. If the specified format has options, the Options button is enabled. Click this button to select options.

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ug.book Page 101 Tuesday, March 16, 2004 1:29 PM

Setting cues automatically

The Auto-Cue feature lets you locate phrases or beats and automatically add them to the cue list. You can also use this feature to remove silence from the beginning and end of a file.

To set cues automatically:

1 Select the general range in which you want to find phrases or beats.

2 Choose Edit > Auto-Cue, and choose one of the following commands:

- Adjust Selection To Phrase selects a phrase within the selected range by adjusting the highlight inward, ignoring any silence before and after the audio. Nothing is added to the cue list.
- Find Phrases And Mark scans the selected range, marking nonsilent ranges as basic cues in the cue list.
- Find Beats And Mark scans the selected range, marking beats as beat cues in the cue list.

To customize Auto-Cue settings:

1 Choose Edit > Auto-Cue > Auto-Cue Settings.

2 Adjust the following options, and click OK:

- Audio Will Be Considered "Silence" When specifies parameters for finding silence. In the Signal Is Below text box, enter the amplitude value (in decibels) you want Adobe Audition to consider as the maximum level for silence. In the For More Than text box, enter the duration (in milliseconds) of this maximum amplitude value.

For very quiet, high-quality audio, enter a lower amplitude value (such as –60 dB). For noisier audio, the value might be much higher (such as –30 dB). Enter a longer duration to keep groups of words together, for example.

- Audio Will Be Considered As Valid When specifies parameters for determining if audio is valid. In the Signal Is Above text box, enter the amplitude value (in decibels) you want Adobe Audition to consider as the minimum level for audio. In the For More Than text box, enter the duration (in milliseconds) of this minimum amplitude.

Enter a longer duration to ignore short periods of undesired audio (like clicks, static, or other noise). However, if the value is too high (above 200 milliseconds), short words may be skipped.

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ug.book Page 102 Tuesday, March 16, 2004 1:29 PM

- Find Levels scans the waveform (or a selected range) to have Adobe Audition automatically determine a good starting point for signal levels. Suggested values appear in the appropriate text boxes.

If these values don't do the job—for example, words or phrases get chopped off—lower the signal level values. Increase the signal level values if not enough silence is removed.

To trim silence from the beginning and ending of a file:

Choose Edit > Auto-Cue > Trim Digital Silence.

If you select the middle of a waveform, this command functions like the normal Trim command, trimming out everything else, in addition to any digital silence in the highlighted range at the endpoints.

Creating play lists

A *play list* is an arrangement of cue ranges that you can play back in any order and loop a specified number of times. The advantage of using a play list is that you can try different versions of an arrangement before you commit to the edits. You create play lists in the Play List window.

To display the Play List window:

Choose Window > Play List. Alternatively, click the Hide/Show Play List button in the toolbar.

To create a play list:

1 If the Cue List window isn't visible, click Show Cue List in the Play List window.

2 In the Cue List window, select the cue ranges you want to add to the play list. (See "Defining and selecting cues" on page 96.)

3 Click Insert Cues in the Play List window. The selection is inserted either before the currently selected item or at the end if nothing is selected.

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ug.book Page 103 Tuesday, March 16, 2004 1:29 PM

To play items in a play list:

Do one of the following:

- To play the entire play list, select the first item in the list, and click Play in the Play List window.
- To play part of the list, select the first item you want to play, and click Play in the Play List window.
- To play a specific item in the list, select that item, and click Autocue in the Play List window.

To change the order of items in a play list:

1. Select the item you want to move.
2. Click Move Up or Move Down.

To set up looping for an item in a play list:

Select the item, and enter a number in the Loop text box. Each item in the play list can loop a different number of times.

To delete items from a play list:

Select the items you want to delete, and click Remove in the Play List window.

Creating and deleting silence

Adobe Audition provides several ways to create silence in and delete silence from a waveform. Creating silence is useful for inserting pauses and removing nonessential noise from an audio file. Removing silence is useful for cleaning up voice prompts and speeding up narratives without affecting the foreground audio.

Creating silence

Adobe Audition provides two ways to create silence in a waveform: by muting part of the existing waveform or by inserting a new duration of silence.

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ug.book Page 104 Tuesday, March 16, 2004 1:29 PM

To mute existing audio data:

1 Select the desired range of audio data.

2 Choose Effects > Silence.

Unlike deleting or cutting a selection, which splices the surrounding material together, applying the Silence effect leaves the duration of the selection intact, and simply zeros the amplitude within it.

To insert a new duration of silence:

1 Place the cursor where you want to insert the silence. Or, if you want to replace part of the existing waveform, select the desired range of audio data.

2 Choose Generate > Silence.

3 Enter the number of seconds of silence you want to generate. Use decimals to enter partial seconds. For example, enter .3 to generate three-tenths of a second of silence.

4 Click OK. Any audio to the right of the cursor is pushed out in time, thereby lengthening the waveform's duration.

Deleting silence

The Delete Silence command detects and removes silence between words or other audio. It's ideal for cleaning up voice prompts and speeding up narratives without affecting the foreground audio.

To delete silence:

1 If you want to delete silence from part of a waveform, select the desired range of audio data. If you don't select a range, Adobe Audition deletes silence from the entire waveform.

2 Choose Edit > Delete Silence.

3 Set the following options as desired, and click OK:

"Silence" Is Defined As Determines what Adobe Audition considers silence. In the Signal Is Below text box, enter the amplitude value (in decibels) you want Adobe Audition to consider as the maximum level for silence. In the For More Than text box, enter the duration (in milliseconds) of this maximum amplitude value.

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ug.book Page 105 Tuesday, March 16, 2004 1:29 PM

For very quiet, low-noise-floor audio, enter a lower amplitude value (such as –60 dB). For noisier audio, you might enter a higher value (such as –30 dB). Enter a longer duration to keep groups of words together, for example.

"Audio" Is Defined As Determines what Adobe Audition considers audio. In the Signal Is Above text box, enter the amplitude value (in decibels) you want Adobe Audition to consider as the minimum level for audio. In the For More Than text box, enter the duration (in milliseconds) of this minimum amplitude value.

Enter a higher duration to ignore short periods of undesired audio (like clicks, static, or other noise). However, if the value is too high (above 200 milliseconds), short words might be skipped.

Find Levels Scans the waveform (or selected range) to have Adobe Audition automatically determine a good starting point for signal levels. Suggested values appear in the appropriate text boxes.

If these values don't do the job—for example, words or phrases are chopped off—lower the signal level values. Increase the signal level values if not enough silence is removed.

Mark Deletions In Cue List Adds each location where silence is removed to the cue list.

Limit Continuous Silence To Specifies the minimum amount of silence (in milliseconds) to keep at all times. Silent ranges shorter than this length aren't removed; silent ranges greater than this length are shortened so that exactly the specified amount of silence remains. Set this value to zero to remove as much silence as possible.

When shortening speech segments, use a setting of 150 milliseconds or so to leave a more realistic, natural sounding pause. Higher values can lead to an artificial sounding pause.

Scan For Silence Now Previews the silence to be removed. This option reports how much silence will be removed and how many sections of silence were found. This option doesn't actually remove silence, but it gives you an idea of what to expect with the current settings when you actually choose the Delete Silence command.

If you have an audio presentation that consists of many cuts separated by silence (such as a reel of several jingles), choose Edit > Delete Silence to make sure that the duration of silence between each cut is the same. For example, if the difference between cuts 1 and 2 is 3.2 seconds, the difference between cuts 2 and 3 is 4.1 seconds, and the difference between cuts 3 and 4 is 3.7 seconds, you can use Delete Silence to make the silence duration between all four cuts exactly 3 seconds.

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ug.book Page 106 Tuesday, March 16, 2004 1:29 PM

Inverting and reversing audio

The Invert effect simply inverts the waveform's samples, making all positive offsets negative and all negative offsets positive. Inverting is useful for lining up amplitude curves when creating loops or pasting. By inverting one channel of a stereo recording, you can also correct out-of-phase channels or create interesting phasing effects. For more information on phase, see the Glossary.

The Reverse effect reverses the order of a waveform's samples so that they play backwards. Reversing is useful for creating special effects.

To invert a waveform:

1 If you want to invert part of a waveform, select the desired range. Otherwise, deselect all audio data to invert the entire waveform.

2 Choose Effects > Invert.

To reverse a waveform:

1 If you want to reverse part of the waveform, select the desired range. Otherwise, deselect all audio data to reverse the entire waveform.

2 Choose Effects > Reverse.

Generating audio

Adobe Audition provides several commands that let you generate new audio data. These commands are different from effects in that they insert new sounds into a waveform rather than alter existing sounds.

Generating DTMF signals

Dual Tone Multi-Frequency (DTMF) signals (also know as touch tones) are used for dialing telephone numbers over phone lines that are capable of responding to touch tone signals. These signals are recommended internationally by the International Telegraph and Telephone Consultative Committee as the signals for push button telephones.

Keep in mind that the DTMF signals generated by telephone push button keypads are different from the Multi-Frequency (MF) tones generated by the telephone network to transmit information. You can use the DTMF Signals command to generate MF tones as well.

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ug.book Page 107 Tuesday, March 16, 2004 1:29 PM

To generate DTMF signals:

1 Place the cursor where you want to insert the signals. Or, if you want to replace part of the existing waveform, select the desired range of audio data.

2 Choose Generate > DTMF Signals.

3 Set the following options as desired, and click OK:

Dial String Specifies the phone number for which you want to generate tones. You can also enter other characters, such as the asterisk (*) and pound (#) symbols, as well as the letters "A," "B," "C," and "D." Entering the pause character (see "Pause Character" in this list) inserts a pause of a defined length.

Tone Time Specifies the milliseconds for which the tones will last. The standard time for DTMF tones is 100 milliseconds.

Break Time Specifies the number of milliseconds of silence between successive tones.

Pause Time Specifies the length that is assigned to the pause character when it is used in the Dial String text box.

Pause Character Specifies which character Adobe Audition interprets as a pause.

DTMF Signals Generates DTMF signals by using combinations of the frequencies 697 Hz, 770 Hz, 852 Hz, 941 Hz and 1209 Hz, 1336 Hz, 1477 Hz, and 1633Hz.

MF Signals (CCITT R1) Generates MF signals (tones that are internal to telephone networks) using paired combinations of the frequencies 700 Hz, 900 Hz, 1100 Hz, 1300 Hz, 1500 Hz, and 1700 Hz.

Custom Specifies the combinations of frequencies to be used in generating signals. Select this option, and then enter values in the Hz text boxes of the keypad.

Amplitude Determines the volume level (as a percentage) of the tones generated, where 100% means maximum volume without clipping.

Twist Specifies how much louder the higher frequency tone is from the lower frequency tone. Enter a value (in decibels) in the Twist text box to increase the volume of the higher frequency tone accordingly.

Reset To DTMF Clears any custom frequency entries and replaces them with the standard DTMF frequency combinations.

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ug.book Page 108 Tuesday, March 16, 2004 1:29 PM

Generating noise

The Noise command lets you generate random noise in a variety of colors. (Traditionally, color is used to describe the spectral composition of noise. Each color has its own characteristics.) Generating noise is useful for creating soothing sounds like waterfalls (perfect for use with the Binaural Auto-Panner function of Adobe Audition) and for generating signals that can be used to check out the frequency response of a speaker, microphone, or other audio system component.

To generate noise:

1 Place the cursor where you want to insert the noise. Or, if you want to replace part of the existing waveform, select the desired range of audio data.

2 Choose Generate > Noise.

3 Set the following options as desired, and click OK:

Color Specifies a color for the noise:

- Brown noise has a spectral frequency of 1/f^2, which means, in layman's terms, that the noise has much more low-frequency content. Its sounds are thunder- and waterfall-like. Brown noise is so called because, when viewed, the wave follows a Brownian motion curve. That is, the next sample in the waveform is equal to the previous sample, plus a small random amount. When graphed, this waveform looks like a mountain range.
- Pink noise has a spectral frequency of 1/f and is found mostly in nature. It is the most natural sounding of the noises. By equalizing the sounds, you can generate rainfall, waterfalls, wind, rushing river, and other natural sounds. Pink noise is exactly between brown and white noise (hence, some people used to call it tan noise). It is neither random nor predictable; it is fractal-like when viewed. When zoomed in, the pattern looks identical to when zoomed out, except at a lower amplitude.
- White noise has a spectral frequency of 1, meaning that equal proportions of all frequencies are present. Because the human ear is more susceptible to high frequencies, white noise sounds very hissy. Adobe Audition generates white noise by choosing random values for each sample.

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ug.book Page 109 Tuesday, March 16, 2004 1:29 PM

Style Specifies a style for the noise:

- Spatial Stereo generates noise by using three unique noise sources and spatially encoding them to seem as if one comes from the left, one from the center, and one from the right. When you listen to the result with stereo headphones, your mind perceives sound coming from all around. To specify the distance from center of the left and right noise sources, enter a delay value in microseconds. About 900 to 1000 microseconds correspond to the maximum delay perceivable. A delay of zero is identical to monaural noise, where left and right channels are the same.
- Independent Channels generates noise by using two unique noise sources, one for each channel. The left channel's noise is completely independent of the right channel's noise.
- Mono generates noise by using a single noise source, with the left and right channels set equally to that source.
- Inverse generates noise by using a single noise source (similar to the Mono option). However, the left channel's noise is exactly inverse of the right channel's noise. When you listen to the result with stereo headphones, your mind perceives sound coming from within your head instead of from somewhere externally.

Intensity Specifies the intensity of the noise on a scale of 2 to 40. At higher intensities, the noise becomes more erratic and sounds harsher and louder.

Duration Determines the number of seconds of noise that Adobe Audition generates.

For very long periods of noise, it's faster to generate a shorter period (say, about 10 to 20 seconds) and delete excess noise at the beginning and end so that the waves start and end at the midpoint. Then, copy and loop (choose Edit > Mix Paste) as many times as needed.

Generating tones

The Tones command lets you create a simple waveform and gives you control over numerous amplitude- and frequency-related settings. Generating tones is a great place to start when you create new sound effects.

To generate tones:

1 Place the cursor where you want to insert the tones. Or, if you want to replace part of the existing waveform, select the desired range of audio data.

2 Choose Generate > Tones.

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ug.book Page 110 Tuesday, March 16, 2004 1:29 PM

3 Do one of the following:

- To create a constant tone, select Lock To These Settings Only. Then, set options as desired, and click OK.
- To create a tone that changes dynamically over time, deselect Lock To These Settings Only. Use the Initial Settings tab to set options for the initial tone, and use the Final Settings tab to set options for the final tone. After you set options, click OK. The tone generated will gradually go from the initial state to the final state.

For more information, search for "Generate Tones options" in Help.

Converting the sample type

A file's sample type determines its sample rate and bit depth, as well as the channel format (whether the waveform is mono or stereo). You can convert the sample type to change any of these attributes.

When you convert the sample type of a file, Adobe Audition directly processes the samples within the file, or resamples the data, so that the audio retains the same pitch and duration as the original file.

About sample rates

During the sampling process, an incoming analog signal is sampled at discrete time intervals. Each interval of analog signal is momentarily observed, and thus, each represents a specific, measurable voltage level. A mathematical conversion generates a digital series of numbers that represent the signal level at that particular point in time. The generated data can be digitally stored or processed.

The *sample rate* is the number of samples (or snapshots) that are taken of an audio signal per second. For example, a sample rate of 44,100 Hz means that 44,100 samples are taken per second. Since sampling is tied directly to the component of time, a system's sample rate determines a system's overall bandwidth—in other words, how many frequencies can be encoded within the audio signal. Higher sample rates generally yield a better quality waveform.

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ug.book Page 111 Tuesday, March 16, 2004 1:29 PM

The most common sample rates for digital audio editing are as follows:

- 11,025 Hz Poor AM Radio Quality/Speech (low-end multimedia)
- 22,050 Hz Near FM Radio Quality (high-end multimedia)
- 32,000 Hz Better than FM Radio Quality (standard broadcast rate)
- 44,100 Hz CD Quality
- 48,000 Hz DAT Quality
- 96,000 Hz DVD Quality

Previewing a different sample rate

The Adjust Sample Rate command lets you preview how an audio file will sound at a different sample rate. This command doesn't convert the sample rate of the audio file—use the Convert Sample Type command to do that. (See "Changing the sample rate" on page 111.)

To adjust the sample rate:

1 Choose Edit > Adjust Sample Rate.

2 Enter a sample rate in the text box, or choose a common sample rate from the list.

3 Click OK.

Note: *Although you can create and edit any sample rate in Adobe Audition, your sound card may not be capable of playing it properly. To check the capabilities of your sound card, choose Options > Device Properties. (See "Setting properties for audio output devices" on page 37.)*

Changing the sample rate

The sample rate of a file determines the overall bandwidth of the waveform (that is, how many frequencies can be encoded within the audio signal). When changing the sample rate, keep in mind that most sound cards support only certain sample rates.

To change the sample rate of a file:

1 Choose Edit > Convert Sample Type. Alternatively, click the Convert Sample Type button in the toolbar.

2 Select a rate from the Sample Rate list, or enter a custom rate in the text box.

3 Drag the Low/High Quality slider to adjust the quality of the sampling conversion.

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ug.book Page 112 Tuesday, March 16, 2004 1:29 PM

Higher values retain more high frequencies (they prevent aliasing of higher frequencies to lower ones), but the conversion takes longer. Lower values requires less processing time but result in certain high frequencies being "rolled off," leading to muffled-sounding audio. Usually, values between 100 and 400 are fine for most conversion needs.

Use higher values whenever you downsample a high rate to a low rate. When upsampling, results from lower values sound almost identical to those from higher values.

4 Select Pre/Post Filter to prevent false frequencies from being generated at the low end of the audio spectrum. Select this option for the best results.

5 Click OK.

Converting between stereo and mono

The Convert Sample Type command is the quickest way to convert a mono waveform into a stereo waveform, and vice versa. (You can also copy the waveform at its current volume directly into one channel or the other.) If you want to place separate waveforms on each channel of a stereo file and mix them at different volume levels, you can use the Mix Paste command instead.

To convert a waveform from mono to stereo, or vice versa:

1 Choose Edit > Convert Sample Type. Alternatively, click the Convert Sample Type button in the toolbar.

2 Select Mono or Stereo.

3 Enter percentages for Left Mix and Right Mix:

- When you convert a waveform from mono to stereo, the Left Mix and Right Mix options let you specify the relative amplitude with which the original mono signal is placed into each side of the new stereo signal. For example, you can place the mono source on the left channel only, the right channel only, or any balance point in between.
- When you convert from stereo to mono, the Left Mix and Right Mix options let you control the amount of signal from the respective channel that will be mixed into the final mono waveform. The most common mixing method is to use 50% of both channels.

4 Click OK.

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ug.book Page 113 Tuesday, March 16, 2004 1:29 PM

To remove all or most of the lead vocals from many stereo music recordings, you can convert a stereo waveform to mono with a Left Mix of 100% and a Right Mix of –100%. Most vocal tracks are positioned in the middle of the stereo field in-phase, so converting the signal so that it's out of phase often greatly reduces or eliminates the vocal track's level.

To create a stereo waveform with different waveforms in each channel:

1. Copy the mono waveform you want to place in the left channel.
2. Create a new file, and choose Edit > Mix Paste.
3. Select Overlap, and deselect Lock L/R. Set the left volume to 100%, set the right volume to 0%, and click OK.
4. Copy the mono waveform you want to place in the right channel.
5. Switch back to the new file you just created, and choose Edit > Mix Paste.
6. This time, set the left volume to 0% and the right volume to 100%. Click OK.

Changing the bit depth

The bit depth of a file determines the dynamic range of the audio. For example, 8-bit resolution provides 256 possible unique volumes, while 16-bit resolution provides 65,536 possible unique volumes. Adobe Audition supports up to 32-bit resolution.

You can raise the bit depth of a file to gain a greater dynamic range, or you can lower the bit depth to reduce the file size. When converting to a lower bit depth, Adobe Audition provides dithering options to help reduce noise and distortion. Although dithering introduces a small amount of white noise, the result is far preferable to the increased distortion that you would otherwise hear at low signal levels. Dithering also lets you hear sounds that would otherwise be masked by the noise and distortion limits of 8-bit audio.

Work at the 32-bit level when processing audio, even if you plan to downsample to 16- or 8-bit for output. You'll achieve better results than at the 16- or 8-bit level. The only time you may want to work at the 16- or 8-bit level is when processing a very large file on a slow computer.

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ug.book Page 114 Tuesday, March 16, 2004 1:29 PM

To change the bit depth of a file:

1 Choose Edit > Convert Sample Type. Alternatively, click the Convert Sample Type button in the toolbar.

2 Select a bit depth from the Resolution list, or enter a custom bit depth in the text box.

3 When you select a lower bit depth, options in the Dither section are enabled. Set the following options as desired, and click OK:

Enable Dithering Enables or disables dithering. If dithering is enabled, Adobe Audition truncates the audio, meaning that unused bits are simply chopped off and discarded. The result gives a crackly effect that fades in and out on very quiet audio passages.

Dither Depth (Bits) Sets the bit amount of dithering to be applied. In general, values of 0.2 to 0.7 give the best results without adding too much noise. Note, however, that as this value is lowered, other unwanted harmonic distortion noise appears. (Lower values are usually okay if you also apply Noise Shaping.)

p.d.f. (probability distribution function) Controls how the dithered noise is distributed away from the original audio sample value.

Usually, Triangular p.d.f. is a wise choice because it gives the best tradeoff among SNR (Signal-to-Noise ratio), distortion, and noise modulation. Triangular p.d.f. chooses random numbers that are generally closer to 0 than to the edges –1 or +1 (that is, the chance of 0 being chosen is twice as great as the chance of 0.5 or –0.5).

p.d.f.	SNR loss	Modulation
Rectangular	3 dB	Yes
Triangular	4.8 dB	No
Gaussian	6 dB	Negligible
Shaped Triangular	4.8 dB	No
Shaped Gaussian	6 dB	Negligible

Noise Shaping Determines the placement when you move noise to different frequencies. The same amount of overall noise is present, but you can place less noise at one frequency at the expense of placing more noise at another. You may also specify that no noise shaping is used.

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ug.book Page 115 Tuesday, March 16, 2004 1:29 PM

Different curves result in different types of background noise. The type of curve to use depends on the source audio, final sample rate, and bit depth. By introducing noise shaping, you may be able to get away with a lower dither depths to reduce the overall background noise level, without introducing a lot of unwanted harmonic noise.

Curve	Sample Rate
Noise Shaping A	44.1 kHz or 48 kHz
Noise Shaping B	44.1 kHz or 48 kHz
Noise Shaping C1	44.1 kHz or 48 kHz
Noise Shaping C2	44.1 kHz or 48 kHz
Noise Shaping C3	44.1 kHz or 48 kHz
Noise Shaping D	44.1 kHz or 48 kHz
Noise Shaping E	44.1 kHz or 48 kHz
Noise Shaping E2	44.1 kHz or 48 kHz
Noise Shaping (44.1KHZ)	44.1 kHz
Noise Shaping (48KHZ)	48 kHz
Noise Shaping (96KHZ)	96 kHz

Note: *In general, there are no really good noise shaping curves for audio at 32 kHz or lower. With audio at those sampling frequencies, try the different curves to see if they help, and just choose the one that sounds the best.*

Converting multiple files to the same sample rate

If you need to make the same conversion on multiple files, you can save time by creating a sample rate conversion preset.

To create a sample rate conversion preset:

1 Choose Edit > Convert Sample Type. Alternatively, click the Convert Sample Type button in the toolbar.

2 Adjust the settings as desired.

3 Click Save As, type a name for the preset, and click OK.

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ug.book Page 116 Tuesday, March 16, 2004 1:29 PM

To apply a sample rate conversion preset:

Choose a preset from the list. The sample type settings change to the settings defined in the preset.

To delete a sample rate conversion preset:

Choose the preset from the list, and click Delete.

Adding file properties

The Wave Properties command opens a tabbed window that lets you add and get information about the active waveform.

Note: *To preserve file properties when you save a file, make sure that you select Save Extra Non-Audio Information.*

To add file information:

1 Choose View > Wave Properties. Alternatively, click the Add Information button in the toolbar.

2 Click the tabs at the top of the dialog box to navigate between different sets of properties.

3 Set the properties as desired, and click OK.

For more information, search for "Adding file properties" in Help.

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ug.book Page 117 Tuesday, March 16, 2004 1:29 PM

Chapter 5: Enhancing and Restoring Audio

Adobe Audition provides many tools you can use to repair audio and improve sound quality. Powerful noise reduction features let you bring old recordings into the digital age. And with a wide range of filter and amplification effects, you can process audio to produce radio-ready sound or unique special effects.

About enhancing and restoring audio

If you need to add brilliance and impact to a new recording, or clean up the sound of an old one, you can use several types of audio enhancement and restoration effects:

- Noise reduction effects that let you remove unwanted hiss, hum, clicks, or pops. (See "Removing noise" on page 125.)
- Filter effects that let you change overall tonal balance, from rumbling bass tones to sparkling highs. (See "Filtering audio" on page 129.)
- Amplitude effects that let you precisely control audio volume for increased radio impact, detailed fade outs, and more. (See "Optimizing amplitude" on page 134.)

All of these effects are available in Edit View, but some don't exist in Multitrack View. Because the two views are linked, however, you can easily overcome this limitation. If a multitrack clip requires noise reduction, for example, simply double-click the clip to process it in Edit View.

About the mastering process

Mastering describes the complete process of restoring and enhancing audio files for a particular medium, such as radio, video, CD, or the Web. In Adobe Audition, you can master either individual audio files in Edit View or groups of files in a batch process. (Batch processing is particularly useful if you plan to burn a group of files to CD. See "About scripting and batch processing" on page 243.)

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ug.book Page 118 Tuesday, March 16, 2004 1:29 PM

The mastering process consists of several stages, which are usually performed in the following order:

1. Analysis To determine the overall frequency and dynamic range of the existing file. (See "Analyzing frequency, phase, and dynamic range" on page 118.)

2. Noise reduction To remove unwanted hiss, hum, clicks, or pops. (See "Removing noise" on page 125.)

3. Equalization To achieve the desired tonal balance. (See "Filtering audio" on page 129.)

4. Compression To maximize perceived volume. (See "Optimizing amplitude" on page 134.)

5. Normalization To ensure that the loudest sounds reach the highest possible level that digital systems allow—0 dBFS. (See "Using the Normalize effect (Edit View only)" on page 137.)

You can reverse the order of the equalization and compression stages, but be aware that the volume of some tonal ranges may be over- or under-emphasized.

Before mastering audio, consider the requirements of the destination medium. If the destination is the Web, for example, the file will likely be played over speakers that poorly reproduce bass sounds. To compensate, you can boost bass frequencies during the equalization stage of the mastering process.

Analyzing frequency, phase, and dynamic range

In Edit View, you can analyze the frequency, phase, and dynamic range of an audio file. These analysis options are particularly helpful when used in conjunction with the many enhancement and restoration effects in Adobe Audition. For example, you can use the Frequency Analysis window to identify problematic frequency bands, which you can then correct with a filter effect. Similarly, you can use the Waveform Statistics dialog box to determine dynamic range and then compress that range with an amplitude effect.

To analyze a multitrack clip, double-click it to access Edit View.

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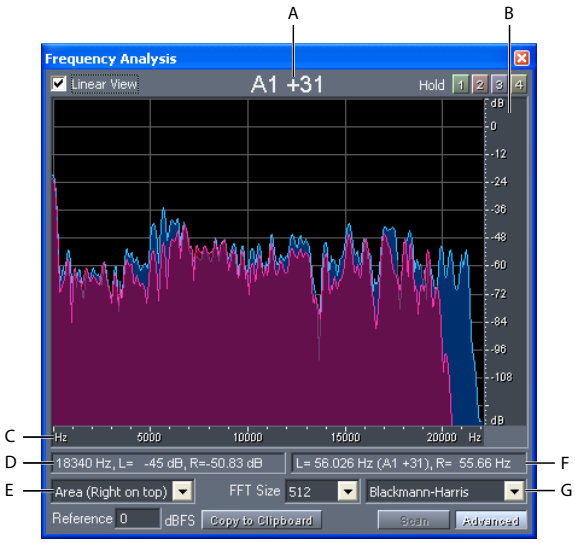
ug.book Page 119 Tuesday, March 16, 2004 1:29 PM

Analyzing frequency range

In Edit View, you can use the Frequency Analysis window to analyze frequency range either statically for a selected area or dynamically during playback. In this window, the horizontal axis represents frequency (measured in Hz), while the vertical axis represents amplitude (measured in decibels).

To zoom in on a particular area of the Frequency Analysis graph, use the horizontal and vertical rulers. See "Zooming graphs for frequency and phase analysis" on page 122.



Frequency Analysis window displaying Advanced options
***A.** Musical note **B.** Vertical ruler **C.** Horizontal ruler **D.** Left status area*
***E.** Display options menu **F.** Right status area **G.** FFT type menu*

To analyze frequency range:

1 In Edit View, select or play a range of the waveform.

2 Choose Window > Frequency Analysis, and set options as desired:

Linear View Sets the graph display to a linear horizontal frequency scale when selected or a logarithmic scale when deselected.

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ug.book Page 120 Tuesday, March 16, 2004 1:29 PM

Hold buttons Take up to four frequency snapshots as a waveform is playing. The frequency outline (which is rendered in the same color as the button clicked) is frozen on the graph and overlaid on other frequency outlines. Up to four frozen frequency outlines may be shown at once. To clear a frozen frequency outline, click its corresponding Hold button again.

Status areas Display frequency and amplitude information directly underneath the graph. The left status area displays the highest frequency of the entire waveform and the maximum amplitude for each channel. The right status area displays the overall frequency (and equivalent musical note) at the center point of the selected range. The numbers beside musical notes indicate keyboard position and variance from standard tuning. For example, A2 +7 equals the second-lowest A on a keyboard tuned 7% higher than normal.

By default, the musical note of the left channel also appears at the top of the window. To hide that note, dock the window, right-click the window handle, and deselect Show Big Notes. For more information, see "Using windows" on page 14.

Display style menu Select from the following graph display options:

- Lines displays amplitude at each frequency with simple lines. The left channel is blue; the right is red.
- Area (Left On Top) also displays lines for amplitude, but this option fills the area beneath the lines in a solid color, smooths out amplitude differences in the same area, and places the left channel in front.
- Area (Right On Top) functions identically to the option above, but places the right channel in front.
- Bars (Left On Top) shows the limitations on analysis resolution by splitting the display into rectangular segments, and places the left channel in front. The higher the FFT size, the greater the analysis resolution, and the narrower the bar.
- Bars (Right On Top) functions identically to the option above but places the right channel in front.

Scan Click this button to scan the highlighted selection and show all frequencies present in that selection.

By default, Adobe Audition analyzes only the center point of a selected range. To analyze the overall frequency of a selected range, click Scan.

For more information, search for "Advanced frequency analysis options" in Help.

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ug.book Page 121 Tuesday, March 16, 2004 1:29 PM

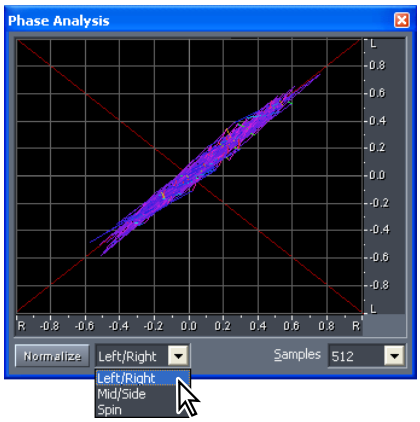
Analyzing phase

In Edit View, you can use the Phase Analysis window to analyze phase either statically for a selected range or dynamically during playback. You should analyze phase only for stereo waveforms, as phase differences don't exist in mono waveforms. Phase analysis can reveal out-of-phase channels, which you can correct with the Invert command. (See "Inverting and reversing audio" on page 106.)

The Phase Analysis window includes a Lissajou Plot graph. By default, this graph displays phase differences between the left and right channels as follows:

- A mono waveform appears as a diagonal line ascending from left to right.
- A right-channel-only waveform appears as a horizontal line.
- A left-channel-only waveform appears as a vertical line.
- A completely out-of-phase stereo waveform appears as a diagonal line descending from left to right.
- A typical stereo waveform appears as many wavy lines descending from right to left.
- A stereo waveform with wide separation appears as many wavy lines extending in all directions.

To zoom in on a particular area of the Phase Analysis graph, use the horizontal and vertical rulers. See "Zooming graphs for frequency and phase analysis" on page 122.



Phase Analysis window with display menu revealed

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ug.book Page 122 Tuesday, March 16, 2004 1:29 PM

To analyze phase:

1 In Edit View, select or play a range of the waveform.

2 Choose Analyze > Show Phase Analysis, and set options as desired:

Normalize Enlarges the phase analysis lines so that they reach the edge of the graph.

Display menu Select from the following options:

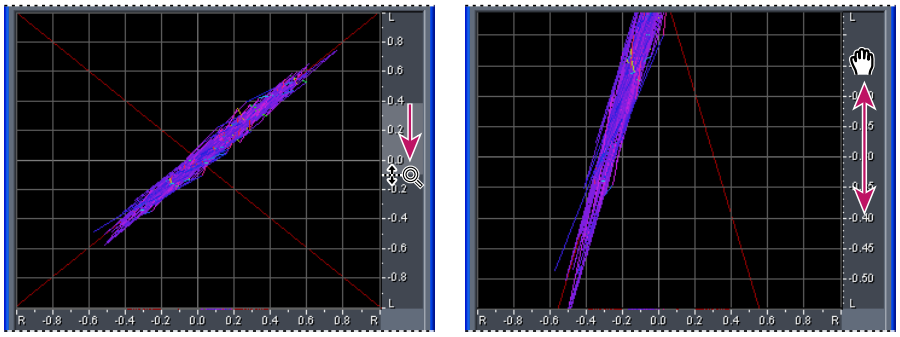
- Left/Right to display the defaults noted in the introduction above.
- Mid/Side to rotate the display to the left by 45 degrees. The horizontal ruler (*x*-axis) plots the side channel [(right - left)/2] while the vertical ruler (*y*-axis) plots the mid channel [(right + left)/2].
- Spin to display the waveform on a phase graph rather than an amplitude graph.

Samples Defines the number of samples displayed concurrently. Higher sample sizes give you more accurate results, but they require much more processing power to be effective. Choose the sample size that best suits your system.

For more information, search for "Advanced phase analysis options" in Help.

Zooming graphs for frequency and phase analysis

In the Frequency Analysis and Phase Analysis windows, you can zoom graphs to analyze frequency and phase in more detail.



Zooming and navigating a Phase Analysis graph

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ug.book Page 123 Tuesday, March 16, 2004 1:29 PM

To zoom in on a graph:

In the vertical or horizontal ruler, right-click and drag the magnifying glass icon.

To navigate a magnified graph:

In the vertical or horizontal ruler, left-click and drag the hand icon.

To zoom out on a magnified graph:

Right-click in the vertical or horizontal ruler, and choose one of the following from the pop-up menu:

- Zoom Out to return to the previous magnification. (This option is available only in the Frequency Analysis window.)
- Zoom Out Full to zoom out completely.

Viewing waveform statistics

In Edit View, you can use the Waveform Statistics dialog box to evaluate a variety of information about audio amplitude. This dialog box contains two tabs, General and Histogram, both of which share an RMS Settings section. The General tab displays numerical text boxes that indicate dynamic range, identify clipped samples, and note any DC offset. The Histogram tab displays a graph that shows the relative prevalence of each amplitude: The horizontal ruler measures amplitude in decibels, and the vertical ruler measures prevalence using the RMS formula.

Use the Histogram tab to identify prevalent amplitudes, and then compress, limit, or normalize them with an amplitude effect. (See "Optimizing amplitude" on page 134.)

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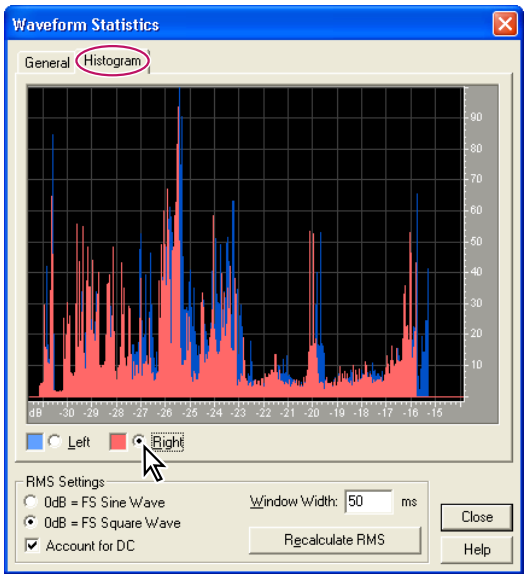
ug.book Page 124 Tuesday, March 16, 2004 1:29 PM



Waveform Statistics dialog box, Histogram tab

To view a waveform histogram:

1 In Edit View, select an audio range.

2 Choose Analyze > Statistics, and click the Histogram tab.

3 Select Left or Right to display either the left or right channel in the foreground.

To view numerical waveform statistics:

1 In Edit View, select an audio range.

2 Choose Analyze > Statistics, and click the General tab.

For more information, search for "Waveform Statistics options" in Help.

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ug.book Page 125 Tuesday, March 16, 2004 1:29 PM

Removing noise

In Edit View, you can use effects in the Noise Reduction menu to reduce background noise and broadband noise without reducing audio quality.

Using the Auto Click/Pop Eliminator effect (Edit View only)

If you need to quickly remove crackle and static from vinyl recordings, first try the Auto Click/Pop Eliminator effect. You can easily select and correct a large area of audio or a single click or pop. This effect provides the same processing quality as the Click/Pop Eliminator effect, but it offers simplified controls and a helpful preview.

To use the Auto Click/Pop Eliminator effect:

1 In Edit View, select an audio range.

2 In the Effects tab of the Organizer window, expand Noise Reduction, and double-click Auto Click/Pop Eliminator.

3 Set the desired options.

For more information, search for "Auto Click/Pop Eliminator options" in Help.

Using the Click/Pop Eliminator effect (Edit View only)

The Click/Pop Eliminator effect detects and removes clicks and pops. Like the Auto Click/Pop Eliminator, this effect is ideal if you want to clean up the sound of vinyl recordings before transferring them to CD or another digital medium. The Click/Pop Eliminator, however, provides a much wider range of controls, letting you highly customize settings for unique audio content.

For this effect, the most important parameters are the Detect and Reject thresholds. (To enable the latter, you must select Second Level Verification.) For Detect thresholds, try settings ranging from 10 for a lot of correction to 50 for very little correction. For Reject thresholds, try settings ranging from 5 to 40. Run Size is the second most important parameter. A setting of about 25 is best for high-quality work. For the highest quality, apply the Click/Pop Eliminator in three successive passes (where each pass is faster than the previous one).

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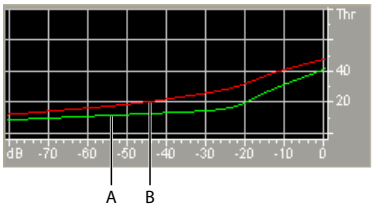
ug.book Page 126 Tuesday, March 16, 2004 1:29 PM



Click/Pop detection graph
A. *Level of detected clicks and pops* ***B.*** *Level of rejected clicks and pops*

To visually identify clicks, zoom in and use Spectral View with a resolution of 256 bands and a window width of 40%. (You can access these settings in the Display tab of the Settings dialog box.) Most clicks appear as bright vertical bars that extend from the top to the bottom of the waveform display.

To use the Click/Pop Eliminator effect:

1 In Edit View, select an audio range.

2 In the Effects tab of the Organizer window, expand Noise Reduction, and double-click Click/Pop Eliminator.

3 Set the desired options.

For more information, search for "Click/Pop Eliminator options" in Help.

Using the Clip Restoration effect (Edit View only)

The Clip Restoration effect repairs clipped waveforms by filling in clipped sections with new audio data. Clipping occurs when the amplitude of a signal exceeds the maximum level for the current bit resolution (for example, levels above 256 in 8-bit audio). Commonly, clipping results from recording levels that are too high. You can monitor clipping during recording or playback by watching the Level Meters; when clipping occurs, the boxes on the far right of the meters turn red.

Visually, clipped audio appears as broad flat areas at the top of a waveform. Sonically, clipped audio is a static-like distortion.

Note: *If you need to adjust the DC offset of clipped audio, first use the Clip Restoration effect. If you instead adjust DC offset first, the Clip Restoration effect won't identify clipped areas that fall below 0 dBFS.*

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ug.book Page 127 Tuesday, March 16, 2004 1:29 PM

To restore clipped audio:

1 In Edit View, select an audio range.

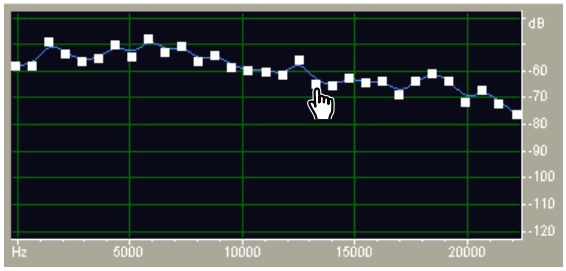
2 In the Effects tab of the Organizer window, expand Noise Reduction, and double-click Clip Restoration.

3 Set the desired options.

For more information, search for "Clip Restoration options" in Help.

Using the Hiss Reduction effect (Edit View only)

The Hiss Reduction effect reduces hiss from sources such as audio cassettes, vinyl records, or microphones. This effect greatly lowers the amplitude of a frequency range if it falls below an amplitude threshold called the *noise floor*. Audio in frequency ranges that are louder than the threshold remain untouched. If audio has a consistent level of background hiss, that hiss can be removed completely.



Using the Hiss Reduction graph to adjust the noise floor

To reduce other types of noise that have a wide frequency range, try the Noise Reduction effect. (See "Using the Noise Reduction effect (Edit View only)" on page 128.)

To reduce hiss:

1 In Edit View, select an audio range.

2 In the Effects tab of the Organizer window, expand Noise Reduction, and double-click Hiss Reduction.

3 Set the desired options.

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ug.book Page 128 Tuesday, March 16, 2004 1:29 PM

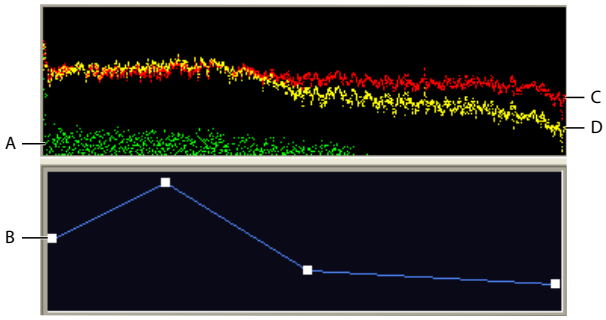
For more information, search for "Hiss Reduction options" in Help.

Using the Noise Reduction effect (Edit View only)

The Noise Reduction effect dramatically reduces background and broadband noise with a minimal reduction in signal quality. This effect can remove a wide range of noise, including tape hiss, microphone background noise, 60-cycle hum, or any noise that is constant throughout a waveform.

The proper amount of noise reduction depends upon the type of background noise and the acceptable loss in quality for the remaining signal. In general, you can increase the signal-to-noise ratio by 5 to 20 dB and retain high audio quality.

To achieve the best results with the Noise Reduction effect, apply it to 16- or 32-bit audio with no DC offset. With 8-bit audio, this effect cannot reduce noise below –45 dB, which is very audible. (To achieve a lower noise floor with 8-bit audio, upsample the file to 16 bits, apply the Noise Reduction effect, and downsample the file back to 8 bits.) With a DC offset, this effect may introduce clicks in quiet passages. (To remove a DC offset, select the Center Wave preset provided by the Amplify/Fade effect.)



Adjusting frequency-specific settings with the Noise Reduction graphs:
***A.** Noise floor **B.** Reduction graph **C.** Original audio **D.** Processed audio*

To reduce noise added by a sound card during recording, start the recording with a second of silence. After recording is complete, use that silence as the Noise Reduction Profile, and then remove it from the complete recording. In some cases, this process can increase dynamic range by 10 dB.

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ug.book Page 129 Tuesday, March 16, 2004 1:29 PM

To reduce noise:

1 In Edit View, select a range that contains only noise and is at least half a second long.

To select noise in a specific frequency range, use the Marquee Selection tool. (See "Selecting audio frequencies in Spectral View" on page 88.)

2 In the Effects tab of the Organizer window, expand Noise Reduction, and double-click Capture Noise Reduction Profile.

3 In the waveform display, select the range from which you want to remove noise.

4 In the Effects tab of the Organizer window, double-click Noise Reduction.

5 Set the desired options.

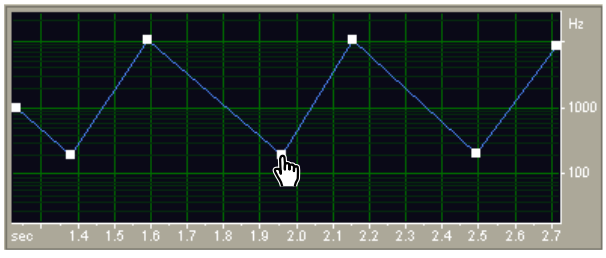
For more information, search for "Noise Reduction options" in Help.

Filtering audio

Filter effects change the frequency content of audio, letting you adjust tonal range to enhance audio or create special effects. (Be aware, however, that significantly boosting a frequency can cause clipping.)

Using the Dynamic EQ effect

The Dynamic EQ effect varies the amount of equalization over time. For example, during the first half of a waveform, you can boost high frequencies; during the second half, you can change the bandwidth of affected frequencies. The Dynamic EQ dialog box provides three areas of controls: Gain, Frequency, and Q (bandwidth).



Frequency graph of the Dynamic EQ effect in Edit View (Rhythmic Sweep preset)

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ug.book Page 130 Tuesday, March 16, 2004 1:29 PM

Dynamic EQ is especially effective as a real-time effect in Multitrack View, where you can use clip envelopes to adjust the Gain, Frequency, and Q parameters.

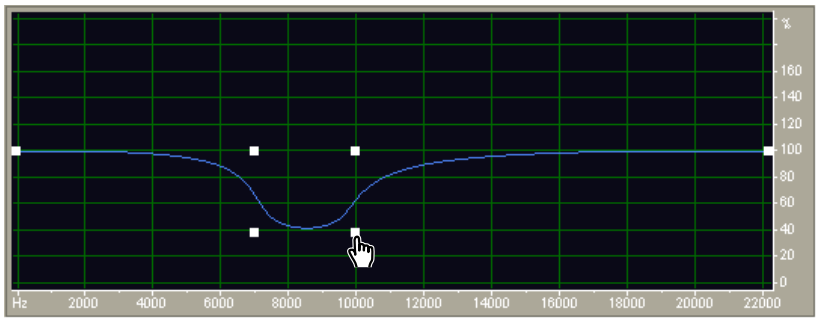
To use the Dynamic EQ effect:

1. Select an audio range (Edit View) or track (Multitrack View).
2. In the Effects tab of the Organizer window, expand Filters, and double-click Dynamic EQ.
3. Set the desired options.

For more information, search for "Dynamic EQ options" in Help.

Using the FFT Filter effect (Edit View only)

The graphic nature of the FFT (Fast Fourier Transform) Filter effect makes it easy to draw curves or notches that reject or boost specific frequencies. This effect can produce broad band-pass filters such as high- and low-pass filters (to maintain high and low frequencies, respectively), narrow band-pass filters (to simulate the sound of a telephone call), or notch filters (to eliminate very narrow frequency bands). The noise level of the FFT Filter effect is lower than that of 16-bit samples, so it introduces no noise when processing audio at 16-bit resolution or lower.



FFT Filter graph (De-Esser preset)

For optimal results, filter 32-bit samples. If the source audio is 8-bit or 16-bit, convert it to 32-bit first, and after you filter it, convert it back to 8-bit with dithering. You'll produce better results than processing at lower resolutions, especially if you perform more than one transform on the audio.

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ug.book Page 131 Tuesday, March 16, 2004 1:29 PM

To use the FFT Filter effect:

1 In Edit View, select an audio range.

2 In the Effects tab of the Organizer window, expand Filters, and double-click FFT Filter.

3 Set the desired options.

For more information, search for "FFT Filter effect options" in Help.

Using the Graphic Equalizer effect

The Graphic Equalizer effect boosts or cuts specific frequency bands and provides a visual representation of the resulting EQ curve. Unlike the Parametric Equalizer, the Graphic Equalizer uses preset frequency bands for quick and easy equalization. The fixed Q settings ensure that no drop outs exist at intermediate frequencies. You can space frequency bands at intervals of one octave, one-half octave, or one-third octave.

The Graphic Equalizer effect is an FIR (Finite Impulse Response) filter, which maintains phase accuracy—unlike an IIR (Infinite Impulse Response) filter, which can introduce phase errors, adding a ringing quality to audio.

To use the Graphic Equalizer effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Filters, and double-click Graphic Equalizer.

3 Set the desired options.

For more information, search for "Graphic Equalizer options" in Help.

Using the Notch Filter effect

The Notch Filter effect removes up to six user-defined frequency bands, in addition to standard telephone DTMF tones. Use this effect to remove very narrow frequency bands, such as a 60 Hz hum, while leaving all surrounding frequencies untouched.

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ug.book Page 132 Tuesday, March 16, 2004 1:29 PM

To use the Notch Filter effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Filters, and double-click Notch Filter.

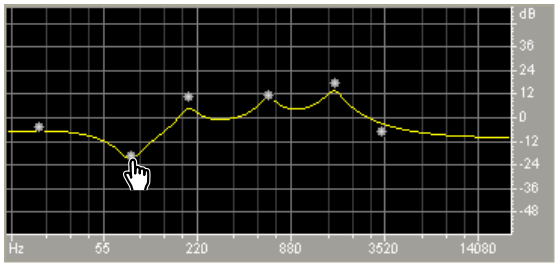
3 Set the desired options.

For more information, search for "Notch Filter options" in Help.

Using the Parametric Equalizer effect

The Parametric Equalizer provides maximum control over tonal equalization. Unlike the Graphic Equalizer, which provides a fixed number of frequencies and Q bandwidths, the Parametric Equalizer gives you total control over frequency, Q, and gain settings. For example, you can simultaneously reduce a small range of frequencies centered around 1000 Hz, boost a broad low-frequency shelf centered around 80 Hz, and insert a 60 Hz notch filter.

The Parametric Equalizer uses second-order IIR filters, which are very fast and provide very precise resolution, even at lower frequencies. For example, you can precisely boost a range of 40 to 45 Hz.



Parametric EQ graph (Old Time Radio preset)

To use the Parametric Equalizer effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Filters, and double-click Parametric Equalizer.

3 Set the desired options.

For more information, search for "Parametric Equalizer options" in Help.

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ug.book Page 133 Tuesday, March 16, 2004 1:29 PM

Using the Quick Filter effect (Edit View only)

The Quick Filter is an 8-band graphic equalizer that you can easily customize to suit many filtering needs. Unlike a standard graphic equalizer, settings for the individual frequency bands interact with nearby frequencies. For example, significantly boosting the level of the highest 22 kHz frequency band moderately boosts the level of lower frequencies. This behavior helps you to quickly and easily enhance audio tone.

To change the equalization setting over time, use different Initial and Final settings. Using this approach, you can create many interesting effects, such as bass-heavy equalization that gradually changes to flat equalization at the introduction of a song.

To use the Quick Filter effect:

1. In Edit View, select an audio range.
2. In the Effects tab of the Organizer window, expand Filters, and double-click Quick Filter.
3. Set the desired options.

For more information, search for "Quick Filter options" in Help.

Using the Scientific Filters effect (Edit View only)

The Scientific Filters effect provides high-order IIR (Infinite Impulse Response) filters for precise band-pass, notch, or high- or low-pass filtering. The most common types of high-order filters are available: Bustle, Butterscotch, Chebychev 1, and Chebychev 2. Each type has different characteristics for filter attenuation and the steepness of transition bands at cutoff points. Butterworth usually provides the best compromise between quality and precision.

On the Scientific Filters graph, one line shows frequency response (measured in decibels), and the other line shows either phase (measured in degrees) or group delay (measured in milliseconds), depending on whether the Phase or Delay option is selected. Increase the graph's display range by selecting Extended Range.

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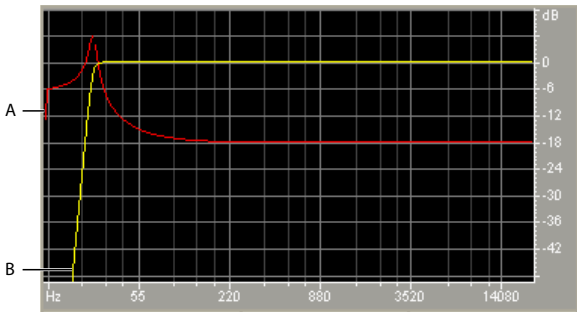
ug.book Page 134 Tuesday, March 16, 2004 1:29 PM



Scientific Filters graph for Butterworth filter (Remove Subsonic Rumble preset)
***A.** Group Delay (milliseconds) **B.** Frequency Response (dB)*

To use the Scientific Filters effect:

1 In Edit View, select an audio range.

2 In the Effects tab of the Organizer window, expand Filters, and double-click Scientific Filters.

3 Set the desired options.

For more information, search for "Scientific Filters options" in Help.

Optimizing amplitude

Amplitude effects let you optimize audio volume for specific mediums such as radio and CD, produce detailed fade outs, and more.

Using the Amplify/Fade effect (Edit View only)

The Amplify/Fade effect produces either constant amplification changes (such as fixed boosts) or precise fades.

Though the Amplify/Fade effect isn't available in Multitrack View, you can use real-time envelopes to accomplish the same task. (See "Automating mixes with clip envelopes" on page 188.)

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ug.book Page 135 Tuesday, March 16, 2004 1:29 PM

To use the Amplify/Fade effect:

1 In Edit View, select an audio range.

2 In the Effects tab of the Organizer window, expand Amplitude, and double-click Amplify/Fade.

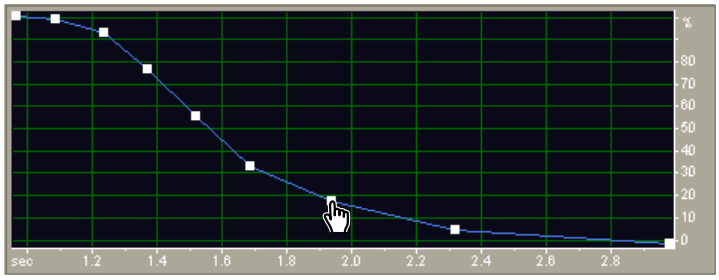
3 Set the desired options.

For more information, search for "Amplify/Fade options" in Help.

Using the Envelope effect (Edit View only)

The Envelope effect lets you precisely control amplitude over time, enabling you to combine a wide range of amplification effects, such as multiple fades and boosts. The top of the Envelope graph represents 100% (normal) amplification; the bottom represents 100% attenuation (silence).

Though the Envelope effect isn't available in Multitrack View, you can use real-time track envelopes to accomplish the same task. (See "Automating mixes with clip envelopes" on page 188.)



Envelope graph (Bell Curve preset)

To use the Envelope effect:

1 In Edit View, select an audio range.

2 In Effects tab of the Organizer window, expand Amplitude, and double-click Envelope.

3 Set the desired options.

For more information, search for "Envelope options" in Help.

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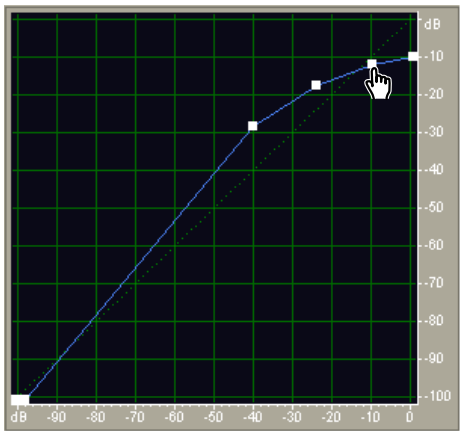
ug.book Page 136 Tuesday, March 16, 2004 1:29 PM

Using the Dynamics Processing effect

The Dynamics Processing effect varies the output level of a waveform based on its input level. You can use this effect to limit or compress dynamic range, producing a consistent level of perceived loudness. You can also expand or gate the signal so that low-level signals are reduced in level, increasing perceived dynamic range, or eliminating signals with noise that falls below a specific threshold.

The Dynamics Processing effect can produce subtle changes that you notice only after repeated listening. When applying this effect in Edit View, use a copy of the original file so you can return to the original audio if necessary.



Dynamics Processing graph (Classic SoftKnee preset)

To use the Dynamics Processing effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Amplitude, and double-click Dynamics Processing.

3 Set the desired options.

For more information, search for "Dynamics Processing options" in Help.

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ug.book Page 137 Tuesday, March 16, 2004 1:29 PM

Using the Hard Limiting effect

The Hard Limiting effect drastically attenuates audio that rises above a defined threshold, leaving audio below the threshold unaffected. This effect is particularly useful for increasing perceived volume because you can amplify audio beyond the digital maximum, 0 dbFS, and you can lower areas that would otherwise be clipped. For example, when you convert from 32-bit to 16-bit audio, particularly loud 32-bit passages can cause 16-bit clipping. To prevent clipping, you can either use the Normalize effect to reduce the amplitude of the entire file (lowering perceived volume), or you can use the Hard Limiting effect to reduce amplitude only for loud passages (increasing perceived volume).

To use the Hard Limiting effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Amplitude, and double-click Hard Limiting.

3 Set the desired options.

For more information, search for "Hard Limiting options" in Help.

Using the Normalize effect (Edit View only)

The Normalize effect lets you set a peak level for a file or selection. When you normalize audio to 100%, you achieve the maximum amplitude that digital audio allows—0 dBFS.

The Normalize effect amplifies the entire file or selection equally. For example, if the original audio reaches a loud peak of 80% and a quiet low of 20%, normalizing to 100% amplifies the loud peak to 100% and the quiet low to 40%.

To apply RMS normalization, you must use the Group Waveform Normalize command. If desired, you can apply that command to only one file. (See "Normalizing groups of files" on page 244.)

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ug.book Page 138 Tuesday, March 16, 2004 1:29 PM

To use the Normalize effect:

1 In Edit View, select an audio range.

2 In the Effects tab of the Organizer window, expand Amplitude, and double-click Normalize.

3 Set the desired options.

For more information, search for "Normalize options" in Help.

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ug.book Page 139 Tuesday, March 16, 2004 1:29 PM

Chapter 6: Applying Stereo, Pitch, and Delay Effects

If you want to add richness and depth to sound, you can use the effects in Adobe Audition to change stereo imagery or simulate acoustic spaces. You can add subtle or psychedelic audio effects, and you can even correct the pitch of an out-of-tune singer.

About using stereo, pitch, and delay effects

Adobe Audition contains a wide variety of effects that let you change stereo imagery, adjust pitch, and add delay (for example, reverb and echo). Dialog boxes for these effects share many common options, such as graphs, spline curves, presets, and previews. For information on these shared options, see "Working with effects" on page 28.

Note: *You can apply effects differently in Edit View and Multitrack View, and some effects dialog boxes have different options in each view. For information on applying effects in Edit View, see "Selecting audio data" on page 87. For information on applying effects in Multitrack View, see "Using real-time effects" on page 185.*

Changing stereo imagery

Adobe Audition lets you change the apparent location, or *stereo imagery,* of sounds coming from the speakers. For instance, you can move a sound from the center to the left or right speaker or even make sounds seem to circle a listener's head.

Note that all of the stereo imagery effects except the Doppler Shifter effect work only on stereo files.

Using the Binaural Auto-Panner effect (Edit View only)

The Binaural Auto-Panner effect spatially lets you designate sound spatially on the left and the right in a seemingly circular pattern over time. In order to spatially encode the sound, either the left or right channel is delayed so that the sounds reach each ear at different times, tricking the brain into thinking they are coming from either side.

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ug.book Page 140 Tuesday, March 16, 2004 1:29 PM

To use the Binaural Auto-Panner effect:

1 In Edit View, select a stereo range.

2 In the Effects tab of the Organizer window, expand Amplitude, and double-click Binaural Auto-Panner.

3 Set the desired options.

For more information, search for "Binaural Auto-Panner options" in Help.

Using the Channel Mixer effect

The Channel Mixer effect alters the left and right balance of a stereo waveform, letting you create new stereo mixes by using the existing right and left channels as input sources. By recombining and inverting the channels, you can create some interesting stereo-imaging effects.

To use the Channel Mixer effect:

1 Select a stereo range.

2 In the Effects tab of the Organizer window, expand Amplitude, and double-click Channel Mixer.

3 Set the desired options.

For more information, search for "Channel Mixer options" in Help.

Using the Pan/Expand effect

The Pan/Expand effect lets you shift the center channel of a stereo waveform. It also lets you expand or narrow the stereo separation of the left and right channels.

Center channel panning uses the *surround* and *center* channels of a stereo recording, where the *surround* channel is the difference of the two original channels, and the *center* channel is the sum of them. You can think of a stereo recording as having four channels (left, right, center, and surround), and this effect lets you pan these channels around. For example, pan hard left to get the original center channel to come out the left speaker and the original surround channel to come out the right. This type of panning can provide added realism to original stereo recordings.

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ug.book Page 141 Tuesday, March 16, 2004 1:29 PM

Expanding works by subtracting or adding differing amounts of right and left channel signals, so sounds occurring on the right or left are cut or boosted. You can alter both of these elements dynamically over time by using the respective graph.

To use the Pan/Expand effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Amplitude, and double-click Pan/Expand.

3 Set the desired options.

For more information, search for "Pan/Expand options" in Help.

Using the Stereo Field Rotate effect

This effect lets you rotate the stereo field of an audio file. The stereo field denotes where in space instruments or other sources are placed within the left and right images of a stereo waveform. By manipulating the Rotation graph, you can affect how the instruments seem to move over time.

To use the Stereo Field Rotate effect:

1 Select a stereo range.

2 In the Effects tab of the Organizer window, expand Amplitude, and double-click Stereo Field Rotate.

3 Set the desired options.

For more information, search for "Stereo Field Rotate options" in Help.

Using the Center Channel Extractor effect

The Center Channel Extractor effect keeps or removes frequencies that are common to both the left and right channels—in other words, sounds that are panned center. Often voice, bass, and lead instruments are recorded this way. As a result, you can use this effect to bring up the volume of the vocals, lead bass, or kick drum or remove any of them from the stereo mix.

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ug.book Page 142 Tuesday, March 16, 2004 1:29 PM

To use the Center Channel Extractor effect:

1 Select a stereo range.

2 In the Effects tab of the Organizer window, expand Filters, and double-click Center Channel Extractor.

3 Set the desired options.

For more information, search for "Center Channel Extractor options" in Help.

Using the Doppler Shifter effect (Edit View only)

The Doppler Shifter effect creates the increase and decrease in pitch we notice when an object approaches and then passes us, such as when a police car passes with its siren on. The frequency of the noise from the siren starts out at a high pitch and tempo, and it lowers as the car passes you. When the car comes toward you, the sound it makes reaches your ears as a higher frequency because each wave crest is actually compressed by the car moving forward. The first crest leaves the car, and by the time the next one leaves, the car has moved forward, reducing the wavelength of the sound and raising its frequency. The opposite happens as the car passes by: The waves are stretched out, resulting in a lower-pitched sound.

To use the Doppler Shifter effect:

1 In Edit View, select an audio range.

2 In the Effects tab of the Organizer window, expand Time/Pitch, and double-click Doppler Shifter.

3 Set the desired options.

For more information, search for "Doppler Shifter options" in Help.

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ug.book Page 143 Tuesday, March 16, 2004 1:29 PM

Using chorus, flanger, and phaser effects

These effects can thicken sound or make it outrageous. They range from the Chorus effect's ability to make a single instrument or vocalist sound like a group playing or singing in unison, to the wilder sounds of the Flanger effect and the phaser effects. Although you can apply them in stereo for the most dimensional results, you can use them with mono sound as well.

Using the Chorus effect

The Chorus effect adds richness as if several voices or instruments are played at once. It's a great way to add a degree of presence to a track. You can use it to give a stereo effect to a mono sample (where the left and right channels are identical) or to add harmony or "thickness" to a vocal track. You can also use it to create some truly out-of-this-world special effects.

Adobe Audition uses a direct-simulation method of achieving a chorus effect, meaning that each voice (or layer) is made to sound distinct from the original by slightly varying the timing, intonation, and vibrato. The Feedback setting lets you add extra detail to the result.

You get better results if you convert mono files to stereo before applying the Chorus effect.

To use the Chorus effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Delay Effects, and double-click Chorus.

3 Set the desired options.

For more information, search for "Chorus options" in Help.

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ug.book Page 144 Tuesday, March 16, 2004 1:29 PM

Using the Flanger effect

Flanging was originally achieved by sending an audio signal to two reel-to-reel tape recorders and then physically slowing down the reels of one machine. The resulting sound has a phase-shifted, time-delay effect, characteristic of psychedelic recordings of the 1960s and 1970s. The Flanger dialog box lets you create a similar result by slightly delaying and phasing a signal at predetermined or random intervals.

To use the Flanger effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Delay Effects, and double-click Flanger.

3 Set the desired options.

For more information, search for "Flanger options" in Help.

Using the Sweeping Phaser effect

Similar to flanging, phasing introduces a variable phase-shift to a split signal and recombines it, creating psychedelic effects first popularized by guitarists of the 1960s. The Sweeping Phaser effect sweeps a notch- or boost-type filter back and forth about a center frequency.

A phase is similar to a flange except that instead of using a simple delay, frequencies are phase-shifted over time. If a phase is used on stereo files, the stereo image can be dramatically altered to create some remarkably interesting sounds.

To use the Sweeping Phaser effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Delay Effects, and double-click Sweeping Phaser.

3 Set the desired options.

For more information, search for "Sweeping Phaser options" in Help.

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ug.book Page 145 Tuesday, March 16, 2004 1:29 PM

Using the Graphic Phase Shifter effect

The Graphic Phase Shifter lets you adjust the phase of a waveform by adding control points to a graph.

To use the Graphic Phase Shifter effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Filters, and double-click Graphic Phase Shifter.

3 Set the desired options.

For more information, search for "Graphic Phase Shifter options" in Help.

Changing pitch

The effects in Adobe Audition let you change the pitch, raising or lowering a person's voice or musical notes. For example, the Pitch Correction effect can correct an out-of-tune vocalist or instrument, and the Stretch effect can stretch or shrink audio without altering pitch or tempo.

Using the Pitch Bender effect (Edit View only)

This effect varies the pitch of the source audio over time. Use the graph to "draw" a tempo to create smooth tempo changes or other effects, such as that of a record or a tape speeding up or slowing down.

To use the Pitch Bender effect:

1 In Edit View, select an audio range.

2 In the Effects tab of the Organizer window, expand Time/Pitch, and double-click Pitch Bender.

3 Set the desired options.

For more information, search for "Pitch Bender options" in Help.

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ug.book Page 146 Tuesday, March 16, 2004 1:29 PM

Using the Pitch Correction effect (Edit View only)

The Pitch Correction effect provides two ways to make pitch adjustments for vocals or solo instrumentation. Automatic mode analyzes the audio content and automatically corrects the pitch based on the key you define, without your having to analyze each note. Manual mode creates a pitch profile that you can adjust note-by-note. You can even over-correct vocals to create robotic-sounding effects.

The Pitch Correction effect detects the pitch of the source audio and measures the periodic cycle of the waveform to determine its pitch. The effect can be used on audio that contains a periodic signal (that is, audio with one note at a time, such as for a saxophone, violin, or vocals). Nonperiodic audio, or periodic audio with a high noise floor, can disrupt the effect's ability to detect the incoming pitch, resulting in incomplete pitch correction.

To use the Pitch Correction effect:

1 In Edit View, select an audio range.

2 In the Effects tab of the Organizer window, expand Time/Pitch, and double-click Pitch Correction.

3 Click the Automatic or Manual tab, and set the desired options.

For more information, search for "Pitch Correct options in Automatic Correction Mode" or "Pitch Correction options in Manual Mode" in Help.

Using the Stretch effect

The Stretch effect lets you change the pitch of an audio signal, the tempo, or both. For example, you can use the effect to transpose a song to a higher key without changing the tempo, or you can use it to slow down a passage without changing the pitch. You can also vary pitch and tempo over the length of the audio, giving the effect of raising and lowering pitch or slowing down and speeding up the tempo.

To use the Stretch effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Time/Pitch, and double-click Stretch.

3 Set the desired options.

For more information, search for "Stretch options" in Help.

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ug.book Page 147 Tuesday, March 16, 2004 1:29 PM

Adding delays and echoes

Delay refers to separating copies of an original signal by some number of milliseconds. *Echoes* are sounds that are delayed far enough in time so that you hear each as a distinct copy of the original sound. Both delays and echoes are a great way to add ambiance to a track where reverb or chorusing might muddy the mix.

Using the Delay effect

Delay can be used to create single echoes, as well as a number of other effects. Delays of 35 milliseconds or more create discrete echoes, while those between 15-34 milliseconds can create a simple chorus or flanging effect. (These results won't be as effective as the actual Chorus or flanging effects in Adobe Audition, as the delay settings are fixed and don't change over time.)

By further reducing a delay to between 1 and 14 milliseconds, you can spatially locate a mono sound (which has the same information in both the left and right channels) so that the sound seems to be coming from the left or the right side, even though the actual volume levels for left and right are identical.

To use the Delay effect:

1. Select an audio range (Edit View) or track (Multitrack View).
2. In the Effects tab of the Organizer window, expand Delay Effects, and double-click Delay.
3. Choose Effects > Delay Effects > Delay.
4. Set the desired options.

For more information, search for "Delay options" in Help.

Using the Dynamic Delay effect

The Dynamic Delay effect lets you change the amount of delay over the length of a waveform. For example, you could set a 2 millisecond delay for the first five seconds of audio, a 20 millisecond delay for the next 15 seconds, a 7 millisecond delay for the next 10 seconds, and so on.

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ug.book Page 148 Tuesday, March 16, 2004 1:29 PM

Dynamic Delay is especially cool when used as a real-time effect in Multitrack View. If you add the dynamic delay (or Dynamic EQ, which has a similar principle) to Multitrack View, you get a new envelope that determines the delay.

To use the Dynamic Delay effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Delay Effects, and double-click Dynamic Delay.

3 Set the desired options.

For more information, search for "Dynamic Delay options" in Help.

Using the Echo effect

This effect adds a series of repeated, decaying echoes to a sound. (For a single echo, use the Delay effect instead.) You can create effects ranging from a Grand Canyon-type"Hello-ello-llo-lo-o" to metallic, clanging drainpipe sounds by varying the delay amount. By equalizing the delays, you can change a room's characteristic sound from one with reflective surfaces (creating echoes that have bright, shiny, high-end sounds) to one that is almost totally absorptive (meaning very few high-end sounds are reflected).

Note: *Make sure that enough silence is at the end of the waveform for the echo to end. If the echo is cut off abruptly before it fully decays, undo the Echo effect, add several seconds of silence by choosing Generate > Silence, and then reapply the Echo effect.*

You can create striking stereo echo effects by setting different left and right values for the Decay, Delay, and Initial Echo Volume controls.

To use the Echo effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Delay Effects, and double-click Echo.

3 Set the desired options.

For more information, search for "Echo options" in Help.

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ug.book Page 149 Tuesday, March 16, 2004 1:29 PM

Using the Echo Chamber effect

The Echo Chamber effect can simulate the ambiance of almost any room. Settings let you specify a virtual room's size and surface characteristics, along with the placement of virtual microphones. The number of echoes is adjustable up to 500,000. Keep in mind that the more echoes you include, the more time Adobe Audition needs to process the effect.

You can create a spatial, stereo expansion effect by setting the virtual microphones farther apart than your actual stereo speakers. For example, if your stereo speakers are 6 feet apart, try setting the left and right virtual microphones 20 or 30 feet apart.

Make sure that enough silence is at the end of the waveform for the echo to end. If the echo is cut off abruptly before it fully decays, undo the Echo effect, add several seconds of silence by choosing Generate > Silence, and then reapply the Echo Chamber effect.

To use the Echo Chamber effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Delay Effects, and double-click Echo Chamber.

3 Set the desired options.

For more information, search for "Echo Chamber options" in Help.

Using the Multitap Delay effect

Multitap Delay can be thought of as a combination of the Delay, Echo, Filter, and Reverb effects. You can create up to 10 delay units, each with its own delay, feedback, and filtering settings.

If one delay unit is placed inside another (as viewed in the chart above the controls), then the echo occurs more than once. As audio travels down the delay line (represented in the chart by the bottom horizontal arrow pointing to the right) portions at any point can be fed back into the delay line anywhere behind the given offset and at any feedback amount, with any high or low cut filter. Experiment to achieve some very interesting effects.

Each delay unit is represented in the graph as a back-leading arrow starting at the Offset and going back the number of milliseconds stated under Delay. A single delay unit is much the same as the Echo function, but with a slightly different filtering setup. (It uses two sliding bands with variable cutoff points instead of eight bands of filtering.)

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ug.book Page 150 Tuesday, March 16, 2004 1:29 PM

To use the Multitap Delay effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Delay Effects, and double-click Multitap Delay.

3 Set the desired options.

For more information, search for "Multitap Delay options" in Help.

Adding reverb

When a sound occurs, it bounces off of different surfaces on its way to your ears. For example, when someone sings in a room, that sound is reflected off the walls, ceiling, and floor, as well as any objects in the room. This reflected sound is called *reverberation,* or *reverb* for short. All these reflected sounds might reach your ears so closely together that you cannot discern them as separate echoes. However, they give an impression of space. With Adobe Audition, you can customize the reverb and replicate a variety of room environments.

For the most precise control of an effects mix in Multitrack View, set real-time reverbs to 0% Original and 100% Reverb. Then, use the effects mixer to control the ratio of dry to reverberant sound.

Using the Full Reverb effect

Full Reverb, like the standard Reverb effect, simulates acoustic space. It's also convolution-impulse-based (like standard Reverb), meaning no ringing, metallic, or other artificial sounding artifacts are present. However, specific resonance can be achieved if desired.

The Full Reverb effect has some unique features, such as Perception, which simulates room irregularities, and source location to place the "singer" off-center, and have the early reflections realistically model their position within rooms that have acoustically desirable dimensions that you can customize. Practically any wall surface or other sound-affecting factors can be simulated by changing the reverb's frequency absorption by using a three-band, parametric-EQ style interface (in the Coloration tab).

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ug.book Page 151 Tuesday, March 16, 2004 1:29 PM

Note: *Because the Full Reverb effect can take longer to process than the other effects, it may not be the best choice for using in real time in Multitrack View. If you use the Full Reverb effect on a track, consider locking the track afterwards so that it doesn't slow down your editing process.*

To use the Full Reverb effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Delay Effects, and double-click Full Reverb.

3 Click the General Reverb tab, the Early Reflections tab, or the Coloration tab, and specify the options you want on each.

Note that when any of the reverb characteristics are modified, a new impulse is built to simulate the environment you specify. (An "impulse" is the data by which every other sample in a waveform is multiplied.) The impulse can be several megabytes in size, requiring more CPU processing power, so you might have to wait a few seconds after clicking Preview for the reverb to be built. The results, however, are much more natural sounding and easier to tailor. Once built, the preview generally runs in real time, and subsequent previews don't require rebuilding the impulse, nor does adjusting any of the Mixing options or selecting Include Direct.

4 Specify any Mixing options you want.

For more information, search for "General Reverb tab options," "Early Reflections tab options," and "Coloration tab options" in Help.

Using the QuickVerb effect

Like Full Reverb and Reverb, the QuickVerb effect adds reverberation to audio to simulate a different acoustic space. It is faster to use, however, because it isn't convolution-based like Full Reverb and Reverb (both of which increase the processing load on your system). As a result, you can make real-time changes more quickly and effectively in Multitrack View, without needing to "lock" effects to a track. For slightly faster processing and more control, you can also use the Studio Reverb effect. For more information, see "Using the Studio Reverb effect" on page 153.

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ug.book Page 152 Tuesday, March 16, 2004 1:29 PM

To use the QuickVerb effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Delay Effects, and double-click QuickVerb.

3 Set the desired options.

For more information, search for "QuickVerb options" in Help.

Using the Reverb effect

The Reverb effect lets you simulate acoustic space, and it consists of both early reflections and echoes that are so closely spaced that they're perceived as a single decaying sound. The Reverb effect is different from the basic Echo effect in that the delays aren't repeated at regularly spaced intervals.

The Reverb effect can create a wide range of high-quality reverb results. It can reproduce acoustic or ambient environments such as a coat closet, a tiled bathroom shower, a concert hall, or a grand amphitheater. The echoes can be spaced so closely together and made to occur at such random times that a signal's reverberated tail decays smoothly over time, creating a warm and natural sound. Alternatively, initial early-reflection delays can be used to give a sense of room size, depending upon the initial delay times.

The difference between the Reverb effect and the Full Reverb effects is that Full Reverb is newer, and it provides more options and better audio rendering. However, you may prefer the older Reverb effect if that's what you're used to using.

Note: *Because the Reverb effect can take longer to process than the other reverb effects, it may not be the best choice for using in real time in Multitrack View.*

To simulate rooms that have both echoes and reverb, use the Echo effect first to establish the "size" of the room sound, and then use the Reverb effect to make the sound more natural. This technique can create a sense of spaciousness in a monophonic signal (one that has been recorded as or converted into a stereo audio file). Even a Total Reverb Length as little as 300 milliseconds can open up the perceived spaciousness of a dry sound (one that was recorded without any effects or reverb).

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ug.book Page 153 Tuesday, March 16, 2004 1:29 PM

To use the Reverb effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Delay Effects, and double-click Reverb.

3 Set the desired options.

For more information, search for "Reverb options" in Help.

Using the Studio Reverb effect

Like Full Reverb, QuickVerb, and Reverb, the Studio Reverb effect adds reverberation to audio to simulate a different acoustic space. It is faster to use than Full Reverb and Reverb, however, because it isn't convolution-based like those effects (both of which increase the processing load on your system). As a result, you can make real-time changes more quickly and effectively in Multitrack View, without needing to lock effects to a track.

Although QuickVerb is not convolution-based and is most similar to Studio Reverb, the latter works slightly faster, has better sound quality, and has more options for better control and tonal variation.

To use the Studio Reverb effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Delay Effects, and double-click Studio Reverb.

3 Set the desired options.

For more information, search for "Studio Reverb options" in Help.

Creating special effects

Effects commands in the Special menu let you introduce processing effects that are both innovative and wild. You can use the Convolution effect to use one waveform to modify another, the Distortion effect to make a waveform sound as if it's coming from an overdriven amplifier or speaker, and the Music effect to create notes and chords sampled from a waveform.

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ug.book Page 154 Tuesday, March 16, 2004 1:29 PM

Using the Convolution effect

The Convolution effect multiplies every sample in one wave (the *impulse*) by the samples contained in another waveform. (An "impulse" is the data by which every other sample in a waveform is multiplied. For instance, if the impulse is a single sample of a full volume "click" sound then the convolution of that impulse with any audio data is just that audio data itself. If that click is at half volume, then the convolution is the audio data at half volume.) In a sense, this effect uses one waveform to "model" the sound of another waveform. The result can be that of filtering, echoing, phase shifting, or any combination of these effects. That is, any filtered version of a waveform can be echoed at any delay, any number of times.

For example, modeling someone saying "Hey" with a drum track (short, full-spectrum sounds such as snares work best) results in the drums saying "Hey" each time they are hit. You can build impulses from scratch by specifying how to filter the audio and what delay rate to apply, or by copying audio directly from a waveform.

With the proper impulses, you can simulate any reverberant space. For example, if you have an impulse of your favorite cathedral, and you convolute it with any mono audio (for which the left and right channels are the same), the result sounds as if that audio were played in that cathedral. You can generate such an impulse by going to the cathedral, standing where you want the audio to seem to be coming from, generating a loud impulsive noise (like a "snap" or "click"), and recording the noise in stereo. If you use this recording as an impulse, convolution with it sounds as if the listener is at the exact location of the recording equipment, and the convoluted audio is at the location of the snap or click.

If several ticks descend in amplitude over time, such as one tick every 100 milliseconds, with each tick half as loud as the previous one, then the resulting convolution with audio has 100 milliseconds between each echo, and each echo is half as loud as the previous one.

To get a feel for Convolution, open and play with some of the sample Impulse (.imp) files that come with Adobe Audition. You can find them in the Imps folder within the folder for Adobe Audition and on the Adobe Audition CD.

Use convolution to sustain a sound for any length of time. For example, the sound of a person singing "aaaaaah" for one second can be turned into thousands of people singing "aaaaaah" for any length of time by using dynamically expanded white noise. Also, to send any portion of an unprocessed "dry" signal back out, simply add a full spectrum echo at 0 milliseconds. The Left and Right volume percentages are the resulting volume of the dry signal in the left and right channels.

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ug.book Page 155 Tuesday, March 16, 2004 1:29 PM

To use the Convolution effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Special, and double-click Convolution.

3 Set the desired options.

For more information, search for "Convolution options" in Help.

Using the Distortion effect

Use this effect to simulate blown car speakers, muffled microphones, or overdriven amplifiers. Have fun making your audio sound really bad or adding fuzz to guitar licks to get that authentic heavy metal sound.

To use the Distortion effect:

1 Select an audio range (Edit View) or track (Multitrack View).

2 In the Effects tab of the Organizer window, expand Special, and double-click Distortion.

3 Set the desired options.

For more information, search for "Distortion options" in Help.

Using the Music effect (Edit View only)

The Music effect lets you use any short selection as a "voice" to synthesize music or harmonize a wave using a particular chord. While this function is far from a complete MIDI authoring studio, it provides a quick and simple way to put a sample to music.

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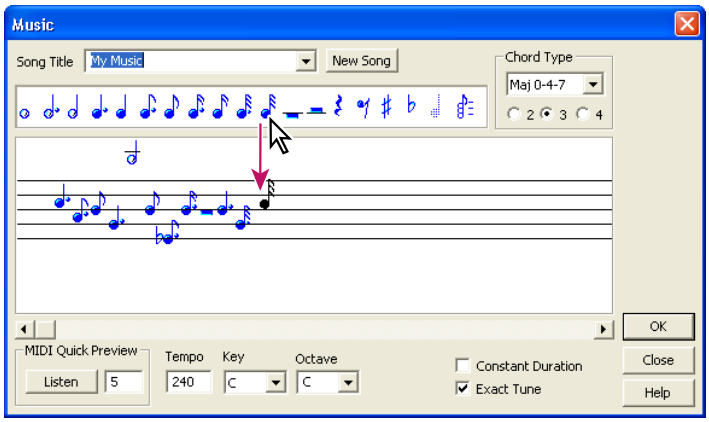
ug.book Page 156 Tuesday, March 16, 2004 1:29 PM



The Music dialog box

To use the Music effect:

1 In Edit View, select the part of the waveform you want to use as a quarter note.

Note: *This selection must be under ten seconds long. If you don't select a range, Adobe Audition uses the data on the clipboard instead. Keep in mind that the clipboard data is filled with the sample automatically after music is generated. Thus, selecting music a second time automatically uses the last sample.*

2 In the Effects tab of the Organizer window, expand Special, and double-click Music.

3 Set the desired options.

For more information, search for "Music options" in Help.

Using multitrack-only effects

Some effects in Adobe Audition are available only in Multitrack View. The Effects menu and the Multitrack category in the Effects tab contain all of these effects. For information about selecting clips and ranges in Multitrack View, see "Selecting ranges in the track display" on page 164.

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ug.book Page 157 Tuesday, March 16, 2004 1:29 PM

Using the Envelope Follower effect (Multitrack View only)

The Envelope Follower effect varies the output level of one waveform based on the input level of another. The amplitude map, or envelope, of one waveform (the analysis wave) is applied to the material of a second waveform (the process wave), resulting in the second waveform taking on the amplitude characteristics of the first. This effect lets you, for example, have a bass guitar line that sounds only when a drum is hit. In this example, the drum waveform is the Analysis wave, and the bass guitar waveform is the Process wave.

In addition to applying an amplitude envelope to a waveform, you can alter the dynamic properties of the resulting signal with a variety of settings to otherwise expand, gate, compress, or limit it.

To use the Envelope Follower effect:

1 In Multitrack View, position the wave clips so that the sections you want to process together are aligned.

2 Select the Hybrid tool or the Time Selection tool I.

3 In the track display, select the range you want to process.

4 Ctrl-click the wave clips you want to process.

Note: *If you select a range by dragging across a clip, that clip is selected by default; if you Ctrl-click the clip, you deselect it.*

5 In the Effects tab of the Organizer window, expand Multitrack, and double-click Envelope Follower.

6 Set the desired options.

For more information, search for "Envelope Follower options" in Help.

Using the Frequency Band Splitter effect (Multitrack View only)

The Frequency Band Splitter lets you take a selected waveform clip (or a highlighted section thereof) and make up to eight copies of it, with each copy assuming a different frequency range of the original. Split points are determined by the crossover frequencies you specify. Each copy of the waveform is placed in its own track in the session window. You can then edit or apply effects to each band separately.

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ug.book Page 158 Tuesday, March 16, 2004 1:29 PM

For example, using the default setting of three bands with crossover values of 800 and 3200 creates three copies of the selected waveform: one with the frequencies of the selected wave from 0 to 800 Hz, one from 800 to 3200 Hz, and one from 3200 to 22050 Hz (or whatever the maximum frequency present is, based on the sample rate).

To use the Frequency Band Splitter effect:

1 In the track display in Multitrack View, select the clip or the range you want to process. (Use the Hybrid tool or the Time Selection tool I to select a range.)

2 Ctrl-click the wave clip.

Note: *If more than one wave clip is selected, the Frequency Band Splitter effect is unavailable. Also, if you select a range by dragging across a clip, that clip is selected by default; if you Ctrl-click the clip, you will deselect it.*

3 In the Effects tab of the Organizer window, expand Multitrack, and double-click Frequency Band Splitter.

4 Set the desired options.

For more information, search for "Frequency Band Splitter options" in Help.

Using the Vocoder effect (Multitrack View only)

A vocoder takes two inputs, usually an instrument and a voice, and modulates one signal (the process signal, usually the instrument) with the other (the control signal, usually the voice). This modulation allows one signal to "control" the other. In the example here, the instrument (the process signal) could be made to "sing" by affecting it with the voice (the control signal).

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ug.book Page 159 Tuesday, March 16, 2004 1:29 PM

To use the Vocoder effect:

1 In Multitrack View, position the wave clips so that the sections you want to process together are aligned.

2 Select the Hybrid tool or the Time Selection tool I.

3 In the track display, select the range you want to process.

4 Ctrl-click the wave clips you want to process.

Note: *If you select a range by dragging across a clip, that clip is selected by default; if you Ctrl-click the clip, you will deselect it.*

5 In the Effects tab of the Organizer window, expand Multitrack, and double-click Vocoder.

6 Set the desired options.

For more information, search for "Vocoder options" in Help.

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ug.book Page 161 Tuesday, March 16, 2004 1:29 PM

Chapter 7: Mixing Multitrack Sessions

In Multitrack View, you can mix together multiple audio files to create layered soundtracks and elaborate musical compositions. Because mixing occurs in real time, it's extremely flexible; during playback, you can adjust mixes and record additional tracks without making any permanent changes. If a mix doesn't sound good next week, or even next year, you can simply remix the original audio files.

About mixing multitrack sessions

In Multitrack View, you can add audio, video, ReWire, and MIDI files to separate tracks of a multitrack session and then mix those tracks together. When you're happy with a mix, you can export a mixdown file for use on CD, the Web, and more.

Multitrack View is a flexible editing environment because mixing occurs in real time and is nondestructive. Because mixing occurs in real time, you can change mix settings during playback and immediately hear the results. For example, you can adjust a track's volume as a session plays to properly blend the track with other tracks. Because mixing is nondestructive, mixing adjustments don't permanently change original source files. For example, you can apply four effects to a track and later remove two effects to create a different sonic texture.

Adobe Audition saves information about mix settings and source files in session (.ses) files. Session files are relatively small because they contain only pathnames to source files and references to mix parameters (such as volume, pan, and effect settings). To more easily manage session files, save them in a unique folder with the source files they reference. If you later need to move the session to another computer, you can simply move the unique session folder. For more information, see "Saving sessions" on page 228.

Note: *Only one session can be open at a time.*

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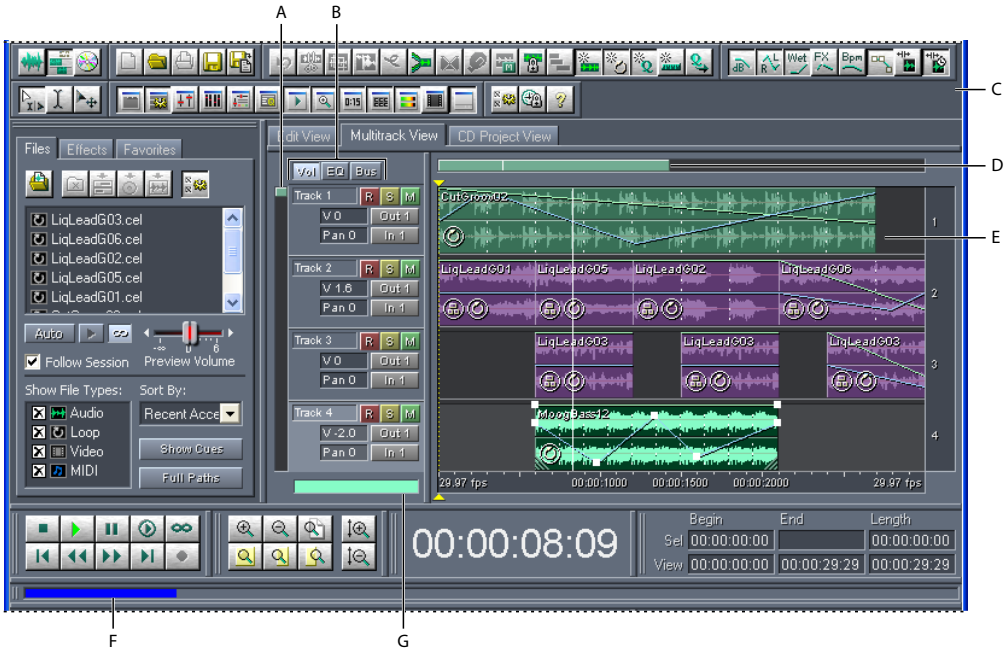
ug.book Page 162 Tuesday, March 16, 2004 1:29 PM

Working with sessions

The Multitrack View work area includes several unique elements that help you mix sessions. On the left, the track controls let you adjust track-specific settings, such as volume and pan. (See "Working with audio tracks" on page 179.) On the right, the track display lets you edit the clips in each track. (See "Working with clips" on page 168.)

For information about elements of the work area that Multitrack View shares with Edit View, see "About the work area" on page 9.



Multitrack View work area
***A.** Vertical scroll bar **B.** Track controls **C.** Toolbar **D.** Horizontal scroll bar **E.** Track **F.** Load Meter **G.** Mix Gauge*

Creating new sessions

When you create a new session, you specify its sample rate. (See "About sample rates" on page 110.) Any files added to the session must share this sample rate. If you try to import a file with a different sample rate, Adobe Audition lets you convert it.

You can base new sessions on the default session, borrowing default settings such as device assignments and master volume levels.

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ug.book Page 163 Tuesday, March 16, 2004 1:29 PM

To convert the sample rate of an existing session, use the Save Session As command and save converted copies of all referenced files. (See "Saving sessions" on page 228.)

To create a new session:

1 Choose File > New Session.

2 Select the desired sample rate.

3 If you want to base the new session on the default session, select Use Default Session. (This option appears only if you've set a default session.)

4 Click OK.

Setting the default session

After you set a default session, it opens when you start Adobe Audition. The default can also serve as a template for new sessions, letting you share settings such as device assignments and master volume levels across multiple sessions.

To set the current session as the default:

1 Choose File > Default Session > Set Current Session As Default.

2 If the session contains clips, click Yes.

To create a new session that uses the default session as a template, see "Creating new sessions" on page 162.

To not use a default session:

Choose File > Default Session > Clear Default Session.

Inserting or deleting time in a session

You can use the Insert/Delete Time command to insert silence into a session or to delete a selected range from the session.

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ug.book Page 164 Tuesday, March 16, 2004 1:29 PM

To insert or delete time in a session:

1 Place the current-time indicator at the desired insertion point, or select the range you want to delete.

2 Choose Edit > Insert/Delete Time, and set the following options:

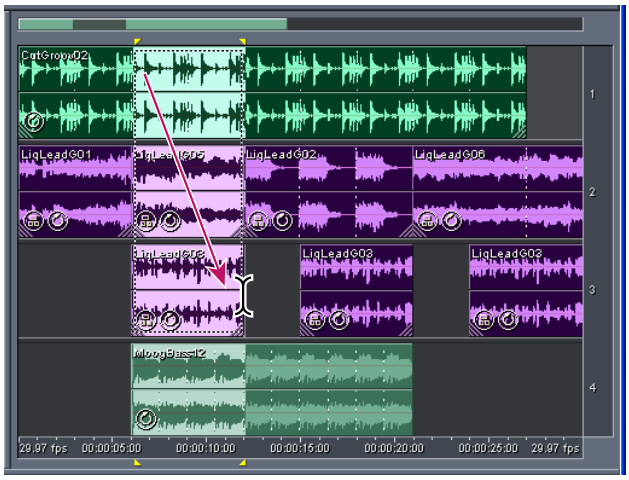
Insert Shifts all material (clips or parts of clips) to the right of the current-time indicator by the amount you specify in the text box. Clips are split if necessary, and the specified amount of silence is inserted.

Delete Selected Time Removes the highlighted area and shifts all clips to the right of the selected region.

You must unlock any locked tracks to insert or delete time in a session. To relock such tracks, click the Lock button in the track controls.

Selecting ranges in the track display

To select ranges in Multitrack View, you can use either the Time Selection tool I or the Hybrid tool . Both tools let you select ranges and clips, but the Hybrid tool also lets you move clips. If you prefer to select ranges separately from moving clips, use the Time Selection and the Move/Copy Clip tools rather than the Hybrid tool. (See "Working with clips" on page 168.)



Simultaneously selecting a range and clips in the track display (Upper three clips are selected, fourth isn't)

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ug.book Page 165 Tuesday, March 16, 2004 1:29 PM

To select a range in the track display:

1 In the toolbar, select either the Hybrid tool or the Time Selection tool.

2 In the track display, do one of the following:

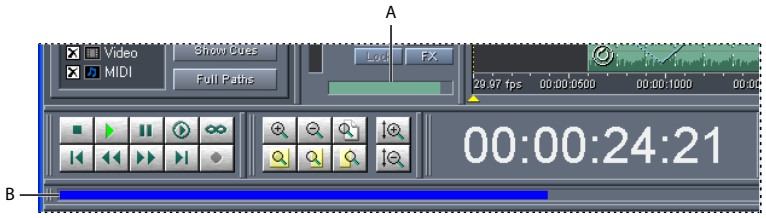
- To select only a range, click an empty area of the track display, and drag left or right.
- To select a range and clips, click a clip, and drag left or right while dragging up or down.

Measuring performance with the Mix Gauge and Load Meter

In Multitrack View, the Mix Gauge and Load Meter help you measure and optimize performance. The Mix Gauge displays the progress of background mixing, a process that Adobe Audition completes whenever you edit a mix (for example, by moving a clip or changing track volume). Background mixing lets you monitor an updated mixdown of a session and is complete when the Mix Gauge reaches 100%. You needn't wait for the Mix Gauge before clicking the Play button, though audio may skip or drop out.

The Load Meter shows the percentage of available CPU power, a particularly important value if you use real-time effects. Unlike the Mix Gauge, the Load Meter indicates a problem if it reaches 100%. At that level, your system will perform erratically because the CPU has no additional processing power. You can reduce CPU load by locking real-time effects. (See "Locking tracks with real-time effects" on page 188).

Multitrack performance depends primarily upon overall system speed, including CPU and hard disk speed. You can optimize multitrack performance on any system, however, by properly configuring multitrack options in the Settings dialog box. (See "Multitrack options" on page 54.)



Performance indicators in Multitrack View
***A.** Mix Gauge **B.** Load Meter.*

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ug.book Page 166 Tuesday, March 16, 2004 1:29 PM

To change background mixing settings:

Right-click the Mix Gauge, and choose any of the following:

- Disable Background Mixing.
- Lower Mix Priority When In Other Applications.
- A Mix Ahead setting to determine how far ahead of the current time Adobe Audition begins mixing. Longer settings allow for faster mix editing, but they might cause drop outs.
- Mix Entire Session to create a new background mix each time you edit a mix.
- A Mix Priority setting to determine the processing priority of background mixing versus other tasks.

Disabling background mixing can improve performance when you need to extensively edit a mix.

To manually start background mixing:

Choose Edit > Refresh Now.

To view or hide the Load Meter:

Choose Window > Load Meter.

Using sessions as SMPTE masters or slaves

By using sessions as SMPTE masters or slaves, you can synchronize the transport controls of Multitrack View with a MIDI sequencing application or an external hardware device, such as a videotape machine. Before using a session as a master or slave, you must set general SMPTE options that apply to all multitrack sessions. (See "Setting up for SMPTE synchronization" on page 40.)

As a SMPTE master, a session generates timecode in the SMPTE time format you select for the timeline. As a SMPTE slave, a session receives timecode generated elsewhere, reporting the following synchronization statuses in the left of the status bar:

- Opened MIDI Input Device when waiting for incoming timecode.
- Synchronizing when establishing synchronization. (Adobe Audition requires about 5 seconds of timecode, known as *preroll*, to establish synchronization.)
- Playback Synchronized when synchronization is established.

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ug.book Page 167 Tuesday, March 16, 2004 1:29 PM

Note: *Adobe Audition sends and receives timecode through the MIDI Out and MIDI In ports of your system. To configure these ports, see "Setting up for SMPTE synchronization" on page 40.*

To use a session as a SMPTE master:

1 Choose Options > SMPTE Master Enable.

2 Select the desired SMPTE time format for the timeline (see "Monitoring time" on page 69).

To use a session as a SMPTE slave:

1 Choose Options > SMPTE Start Offset, click Format, and select the desired SMPTE time format.

2 Enter the desired start point in the SMPTE Start Time Offset box, and then click OK. (This option defines Adobe Audition's start point; it doesn't offset incoming timecode.)

Note: *If you chose the SMPTE Drop time format, the offset must compensate for dropped frames. For example, you must enter 1:00:02 to achieve an offset of 1:00:00.*

3 Choose Options > SMPTE Slave Enable.

Setting advanced session properties

In the Advanced Session Properties dialog box, you can adjust session-specific mixing, tempo, and metronome settings. You can also add session notes, which can help you recall details about a session or communicate those details to someone else.

To set loop-related session properties, use the Session Properties window. See "Setting the tempo, time signature, and key for sessions" on page 204.

To set advanced session properties:

1 Choose View > Advanced Session Properties.

2 Set options as desired, and click OK.

For more information, search for "Setting advanced session properties" in Help.

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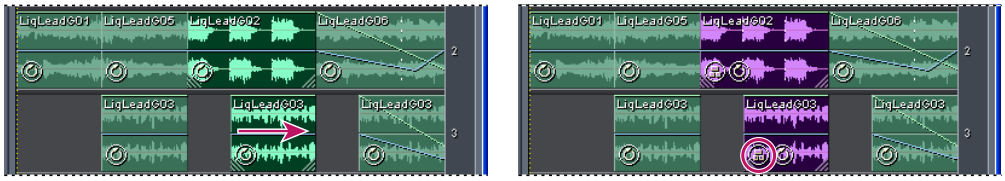
ug.book Page 168 Tuesday, March 16, 2004 1:29 PM

Working with clips

When you insert an audio, MIDI, or video file in Multitrack View, the file becomes a clip on the selected track. You can easily move clips to different tracks or timeline positions. You can also edit clips nondestructively, trimming their start and end points, crossfading them with other clips, and more.

To work with clips in the track display, you can use either the Hybrid tool, which lets you move clips and select ranges, or the Move/Copy Clip and Time Selection tools, which separate these tasks. (See "Selecting ranges in the track display" on page 164.)



Aligning and grouping two clips

Selecting and moving clips

To move a clip or change its properties, you must select it. You can select either individual clips or all clips in a track or session.

To select an individual clip:

Click the clip in the track display.

To select all clips in a track:

1 Select the track.

2 Choose Edit > Select All Clips In Track [number].

If space exists between clips, double-click that space to quickly select all clips in a track.

To select all clips in a session:

Choose Edit > Select All Clips.

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ug.book Page 169 Tuesday, March 16, 2004 1:29 PM

To move selected clips:

1 Select the Move/Copy Clip tool in the toolbar.

2 Drag the clips.

If you prefer, select the Hybrid tool, and then right-click and drag the clips.

Grouping clips

You can group clips to more efficiently organize, edit, and mix a session. For example, you can group guitar clips together to easily identify, select, and move them. Grouped clips appear with the group icon and in a different color than ungrouped clips.

Changes to clip mute and lock properties affect all audio clips in a group. See "Setting audio clip properties" on page 174.

To group clips:

1 Hold down the Ctrl key, and click each clip you want in the group.

2 Choose Edit > Group Clips. Alternatively, right-click any clip in the group, and choose Group Clips.

To ungroup clips:

Select any clip in the group, and choose Edit > Group Clips. Alternatively, right-click any clip in the group, and deselect Group Clips.

To change the color for a group:

1 Select any clip in the group, and choose Edit > Group Color. Alternatively, right-click any clip in the group, and choose Group Color.

2 Select a color, and click OK.

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ug.book Page 170 Tuesday, March 16, 2004 1:29 PM

Aligning clips

You can align the left or right edges of multiple clips, giving them the same start or end point.

To align clips:

1 Hold down Ctrl, and select the clips.

2 Choose Edit > Align Left or Edit > Align Right.

Note: *Because the relative position of grouped clips is fixed, you must ungroup them to align them.*

Snapping clips to loop endpoints and other clips

Snapping lets you quickly align clips with loops and other clips. If snapping is enabled, both dragged clips and the current-time indicator snap to loop endpoints and clip edges.

The procedure in this section describes snapping options that are unique to Multitrack View. For information about snapping options that Multitrack View shares with Edit View, see "Snapping" on page 91.

To set snapping options for clips:

Choose Edit > Snapping, and choose from the following options:

Snap To Clips Causes clips to snap to the beginning or end of other clips.

Snap To Loop Endpoints Causes clips to snap to the beginning or end of loops.

While you drag a clip, a white line appears in the track display when snapping points meet. For example, if Snap To Clips is selected, the white line appears when a clip is aligned with the beginning or end of another clip.

Editing audio and MIDI clips

You can edit audio and MIDI clips to suit the needs of a mix. Because Multitrack View is nondestructive, clip edits are impermanent; you can return to the original, unedited clip at any time. If you want to permanently edit an audio clip, however, you can quickly open the source file in Edit View.

You can edit audio and MIDI clips in many different ways. After selecting a range of a clip, you can cut out that range or trim the clip to it. You can adjust the edited boundaries of a clip, revealing or hiding more of it. You can also slip edit a clip to move its contents but not its boundaries.

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ug.book Page 171 Tuesday, March 16, 2004 1:29 PM

Using the Adjust Boundaries command to reveal more of a previously edited clip

Though the procedures in this section mention toolbar buttons for clip editing commands, you can also access these commands from the Edit menu or the clip context menu. For example, you can choose Edit > Trim instead of clicking the Trim To Selection button.

To edit a clip with a selected range:

1 In the toolbar, click the Time Selection tool or the Hybrid tool.

2 Drag across the clip to select both it and a range.

3 In the toolbar, do one of the following:

- To trim the clip to the range, click the Trim To Selection button.
- To cut the range from the clip, click the Cut Wave(s) Out Of Selection button. (Alternatively, press Delete.)
- To adjust clip edges to the range, click the Adjust Waveform Boundaries To Selection button. (To reveal more of a previously edited clip, extend the range beyond the current clip edges.)

To edit clip edges by dragging:

1 In the toolbar, click the Clip Edge Dragging button.

2 In the track display, position the cursor over the left or right edge of the clip. The edge-dragging icon appears. (If instead the time stretch icon appears, position the cursor above the corner handle.)

3 Drag to edit clip edges.

To slip edit a trimmed or looped clip:

1 In the toolbar, click the Move/Copy Clip tool or the Hybrid tool.

2 Hold down Alt, and right-click drag across the clip.

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ug.book Page 172 Tuesday, March 16, 2004 1:29 PM

To return to the full, original version of a clip:

Select the clip, and choose Edit > Full. Alternatively, right-click the clip, and choose Full.

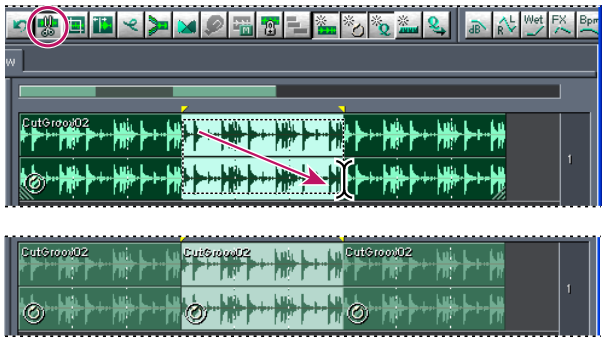
The Full command doesn't apply to loops; instead, adjust clip boundaries by dragging them.

To edit the source file for an audio clip in Edit View:

Double-click the clip.

Splitting and rejoining audio and MIDI clips

The Split command functions similarly to a traditional tape splice; it cuts audio and MIDI clips into parts. When a clip is split, each part becomes a new clip that can be independently moved or deleted. Splitting is nondestructive, so you can rejoin split clips with the Merge/Rejoin Split command.



Selecting a range and splitting one clip into three independent clips

To split a clip:

1 In the toolbar, click the Time Selection tool or the Hybrid tool.

2 Do either of the following:

- To split the clip in two, click where you want the split to occur.
- To split the clip into three, drag across it to specify two split points (one at the beginning of the selection; one at the end).

3 In the toolbar, click the Split Clip button.

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ug.book Page 173 Tuesday, March 16, 2004 1:29 PM

To rejoin split clips:

1 In the toolbar, click the Move/Copy Clip tool or the Hybrid tool.

2 Position the clips beside each other on the same track.

3 Right-click one of the clips, and choose Merge/Rejoin Split.

Copying audio and MIDI clips

You can create two types of copied audio clips: reference copies that share source files and unique copies that have independent source files. You can create only reference copies of MIDI clips. For audio clips, the type of copy you choose depends upon the amount of available disk space and the nature of destructive editing you plan to perform in Edit View.

Reference copies consume no additional disk space, letting you simultaneously edit all instances by editing the original source file. (For example, you can add the Flanger effect to the source file in Edit View and automatically apply the effect to all 30 referenced copies in a session.)

Unique copies have a separate audio file on disk, allowing for separate editing of each version in Edit View. (For example, you can add destructive effects to the version in an introduction while leaving the version in a verse dry).

To copy a clip:

1 Click the Move/Copy Clip tool in the toolbar.

2 Right-click and drag the clip.

3 Release the mouse button, and choose either of the following from the pop-up menu:

- Copy Reference Here
- Copy Unique Here

If you prefer, copy clips with the Hybrid tool. To copy a reference clip, hold down Shift and right-click drag. To copy a unique clip, hold down Ctrl and right-click drag.

Repeating audio and MIDI clips

With the Clip Duplicate command, you can duplicate repetitions of a clip in a track without consuming additional disk space. You can also specify the spacing between each repetition.

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ug.book Page 174 Tuesday, March 16, 2004 1:29 PM

To copy a clip to a different track or to irregular positions in the current track, see "Copying audio and MIDI clips" on page 173.

To repeat a clip:

1 Select the clip, and choose Edit > Clip Duplicate.

2 Set the following options:

Duplicate Clip Specifies the number of times to duplicate the clip.

Spacing Determines the spacing between each duplicated clip:

- No Gaps—Continuous Looping places each duplicate directly after its preceding clip, for a continuous loop.

For a more flexible method of looping, adjust a clip's loop properties. See "About loops" on page 197.

- Evenly Spaced defines the spacing between each clip according to the time display format. This value defaults to the length of the selected clip, producing the same effect as the No Gaps option. Enter a greater value to place space between each clip, or enter a lesser value to overlap clips.

To repeat a clip such as a drum hit at every other beat in a song, set the time format to Bars And Beats. (See "Monitoring time" on page 69.) If the clip's start and end points don't align properly with beats, trim the clip in Edit View by using Edit > Find Beats.

Setting audio clip properties

In the Audio Clip Properties window, you can change settings such as volume, pan, and color for audio clips. Clip settings for volume, pan, and mute are independent from similar track controls.

You can also lock clips in time and lock them for play only. If a clip is locked in time, you can move it up or down to another track, but you can't move it right or left to a new timeline position. If a clip is locked for play only, you can record in the remainder of the track without recording over the clip.

You can directly access many audio clip properties from the Edit menu or the clip context menu (for example, choose Edit > Adjust Audio Clip Volume).

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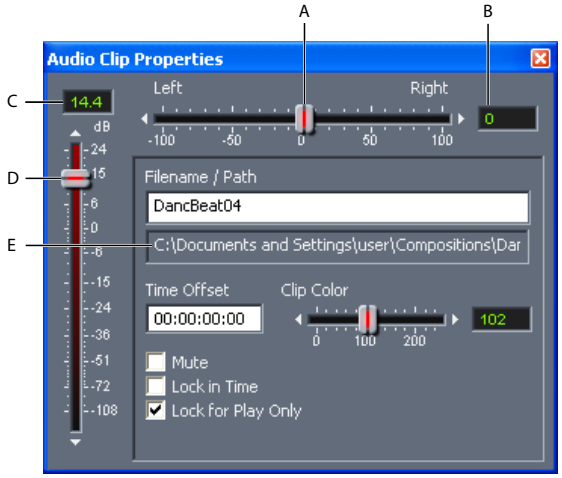
ug.book Page 175 Tuesday, March 16, 2004 1:29 PM



Audio Clip Properties window
***A.** Pan slider **B.** Pan text box **C.** Volume text box **D.** Volume slider*
***E.** Pathname for source file*

To change the properties of an audio clip:

1 Right-click the clip, and choose Audio Clip Properties.

2 Do any of the following:

- To change volume, pan, or color, drag the volume, pan, or color slider to the desired position.
- To lock the clip in time, select Lock In Time. A lock icon appears on the clip.
- To lock the clip for play only, select Lock For Play Only. If the containing track is record-enabled, the clip remains the same color; other clips in the track turn red.
- To mute the clip, select Mute.
- To move the clip to a new timeline position, enter a start time in the Time Offset text box.
- To change the clip name, type in the Filename text box. (When you save the session, Adobe Audition prompts you to save a copy of the source file with the new clip name.)

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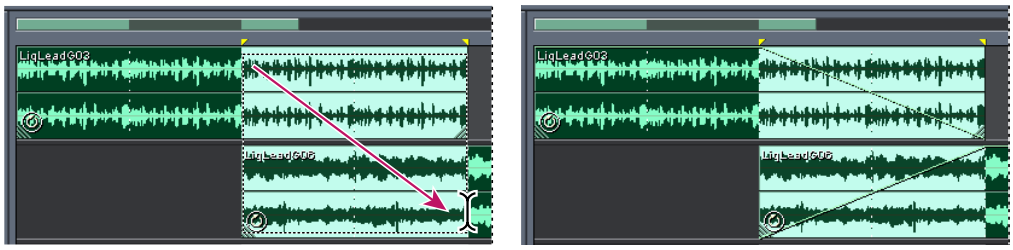
ug.book Page 176 Tuesday, March 16, 2004 1:29 PM

Crossfading audio clips

You can crossfade audio clips to transition smoothly from the end of one clip to the beginning of another. Crossfades consist of a fade out and a fade in over a transition region. To create a smooth transition, select a transition region that starts before the end of the first clip and extends beyond the beginning of the second clip.

The fade curves created with Crossfade commands are volume envelopes, which you can edit. See "Automating mixes with clip envelopes" on page 188.



Selecting a range and two clips, and applying a linear crossfade

To crossfade two clips:

1. Place the clips on separate tracks.
2. Position the clips so the end point of the first overlaps the start point of the second.
3. Across the overlapping area, select a transition region for the crossfade.

To precisely place the start and end points for the crossfade at clip start and end points, choose Edit > Snapping > Snap To Clips.

4. Ctrl-click both clips.
5. Choose Edit > Crossfade, and then choose one of the following:

- Linear to produce an even crossfade.
- Sinusoidal to produce a crossfade with a curved, sine-like slope.
- Logarithmic In to fade in logarithmically, producing a steeper slope at the end of the fade.
- Logarithmic Out to fade out logarithmically, producing a steeper slope at the beginning of the fade.

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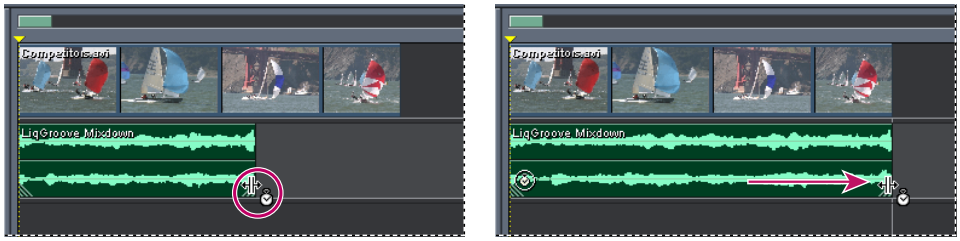
ug.book Page 177 Tuesday, March 16, 2004 1:29 PM

Time stretching audio clips

Time stretching lets you change the length of an audio clip without changing its pitch. This technique is particularly helpful for fitting audio clips to video scenes or layering clips for sound design. You can quickly time stretch a clip either by dragging or setting time stretch properties. When you time stretch by dragging, Adobe Audition analyzes a clip's contents and attempts to select the most natural sounding time-stretch method. When you set properties for time stretching, you also specify which method of time stretching to use.

Like other features in Multitrack View, time stretching is nondestructive, so you can disable it at any time.

Note: *Time stretching changes the tempo of a clip. If you time stretch a loop-enabled clip, it won't match the session tempo.*



Dragging to time stretch a clip

To time stretch a clip by dragging:

1 In the toolbar, click the Clip Time Stretching button .

2 Select the clip, and then position the cursor over the clip's bottom left or right handle—the time stretch icon appears.

3 Drag the handle to lengthen or shorten the clip.

To temporarily enter time stretching mode, hold down Ctrl, and drag a clip handle.

To set time stretch properties:

1 Right-click the clip, and choose Clip Time Stretch Properties.

2 Select Enable Time Stretching, and enter a percentage in the Time Stretch text box.

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ug.book Page 178 Tuesday, March 16, 2004 1:29 PM

3 Choose one of the following time stretching options from the pop-up menu, set related options, and then click OK:

Time-Scale Stretch Stretches the clip without affecting pitch. This method is most commonly used for melodic instruments, like piano, bass, and guitar. Because this method bases the stretch on the actual length and duration of the file, use it only to stretch audio that doesn't have well-defined beats, like a synth pad or sustained string section.

Resample (Affects Pitch) Speeds or slows the playback of a clip to fit the new length without maintaining pitch. This setting is commonly used in R&B and hip hop to achieve exaggerated stretching and compressing of drum tracks, creating a lo-fi sound. This setting also works well for vocals, allowing subtle to radical changes in timbre.

Beat Splice Stretches the clip based on beats detected within the file. This setting works only on clips that have very sharp, transient sounds, like drums. If the waveform already has beat markers, select Use File's Beat Markers to use them. Otherwise, select Auto-Find and adjust the default values as needed.

Hybrid Uses the current Time-Scale Stretch settings when you shorten the clip, and uses the current Beat Splice settings when you lengthen it.

To disable time stretching:

1 Right-click the time-stretched clip, and choose Clip Time Stretch Properties.

2 Deselect Enable Time Stretching.

Inserting empty audio clips

You can insert empty audio clips as placeholders for audio you plan to record later. This technique is particularly helpful when combined with the Punch In command. (See "Recording audio in Multitrack View" on page 73.)

To insert an empty audio clip:

1 Select a range in the track display.

2 Choose Insert > Empty Audio Clip, and then choose one of the following:

- In Current Track (stereo)
- In Current Track (mono)
- In All Record-Armed Tracks

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ug.book Page 179 Tuesday, March 16, 2004 1:29 PM

Revealing hidden clips

If tracks contain overlapping clips, you can reveal hidden clips throughout a session.

To reveal hidden clips:

Choose Edit > Check for Hidden Clips.

Removing and destroying clips

You can remove selected clips from a session and keep their source files available in the Insert menu and in Edit View. Alternatively, you can destroy selected clips to remove them from a session and close their source files.

To remove selected clips:

Choose Edit > Remove Clips.

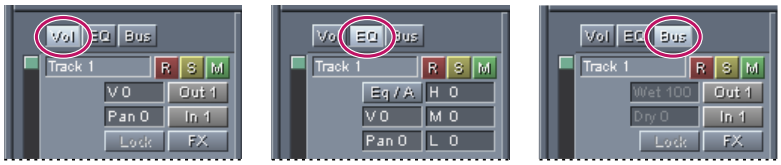
To destroy selected clips:

Choose Edit > Destroy Clips.

Working with audio tracks

You can record and mix up to 128 tracks in Adobe Audition, and each track can contain as many clips as you need—the only limit is hard disk space. The track controls appear to the left of the track display, and you can resize these controls to be as wide or narrow as you wish. On the Vol, EQ, and Bus tabs, you can access different sets of controls for volume, equalization, and bus properties. Though the wide variety of track controls may seem intimidating at first, the controls for each track are identical, so if you learn one, you've learned them all.

For information about recording on audio tracks, see "Recording audio in Multitrack View" on page 73.



Track controls on the Vol, EQ, and Bus tabs

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ug.book Page 180 Tuesday, March 16, 2004 1:29 PM

Using the Track Properties window

In the Track Properties window, you can adjust several settings for the selected track, including volume, pan, output device, and bit depth. Though you can quickly access most of these options in the track controls, the Track Properties window offers channel and bit depth menus, and visual sliders for volume and pan.

To use the Track Properties window:

1 Select the track, and then choose Window > Track Properties.

2 Set the desired options.

For more information, search for "Track Properties options" in Help.

Setting track name, volume, and pan

In the track controls, you can name tracks to identify their contents (for example, "Drums"). You can also specify volume and pan settings.

To change track volume and pan over time, use track envelopes. (See "Automating mixes with clip envelopes" on page 188.)

To name a track:

In the track controls, type in the name text box.

Name text box in the track controls

To change track volume or pan:

In the Volume (V) or Pan text box of the track controls, drag to scroll through values.

To change these settings with a slider, right-click the Volume or Pan text box.

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ug.book Page 181 Tuesday, March 16, 2004 1:29 PM

Soloing and muting tracks

You can solo tracks to hear them separately from the rest of a mix. Conversely, you can mute tracks to silence them in a mix.

To solo a track:

In the track controls, click the Solo button S.

To solo multiple tracks, hold down Ctrl and press their Solo buttons.

To mute a track:

In the track controls, click the Mute button M.

Specifying track input and output devices

Using the In and Out buttons in the track controls, you can specify input and output devices for each track. The text on these buttons changes to reflect the device you specify.

When you specify an output device, you can specify either a hardware output or a bus output. Bus outputs let you create submixes of selected tracks (for example, drum tracks), which you can then route to a hardware output. (See "Using the Bus Mixer" on page 193.)

On the EQ tab of the track controls, the In and Out buttons are hidden by default. To reveal these buttons, increase the width of the track controls by dragging the right border.

To specify an input device for a track:

1 In the track controls, click the In button.

2 From the Device Type menu, choose the device type.

3 From the list box, select the input device.

4 In the Input Options section, specify the channel and bit depth. (To apply these input options to all tracks, select Same For All Tracks.)

To specify an output device for a track:

1 In the track controls, click the Out button.

2 Select a hardware output from the Devices list or a bus output from the Busses list.

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ug.book Page 182 Tuesday, March 16, 2004 1:29 PM

The list of devices is determined by the devices you designate in the Device Order dialog box. (See "Designating which devices you want to use" on page 36.)

Setting track channel and bit depth

To set the channel and bit depth for a track, use the Track Properties dialog box.

Track Properties dialog box with channel menu revealed and bit-depth menu highlighted

To set channel and bit depth for a track:

1. Select the track, and then choose Window > Track Properties.
2. Choose an option from the channel and bit depth menus.

Equalizing tracks

You can equalize audio tracks by using either the track controls or the Track Equalizers window. The track controls provide quick access to commonly used equalization settings; the Track Equalizers window provides access to more precise and sophisticated controls.

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ug.book Page 183 Tuesday, March 16, 2004 1:29 PM

In the EQ tab of the track controls, the track equalization text boxes show the current low-, mid-, and high-frequency equalization. You can drag across these text boxes to change equalization settings. To switch between two banks of equalization settings, you can use the Eq/A or Eq/B button. For example, you can adjust settings for the Eq/A bank, and then click the button to access the default, unequalized settings for the Eq/B bank. However, if you change settings while Eq/B is active, those settings are preserved. This functionality lets you compare any two settings.

In the Track Equalizers window, you can specify the center frequency and Q range for the low, middle, and high bands. Then, you can use a graph to visually adjust equalization settings.

Note: *In the track controls, track equalization text boxes appear in the EQ tab by default. To reveal these fields in other tabs, increase the width of the track controls.*

Switching between Eq/A and Eq/B settings

To equalize a track by using the track controls:

1 On the EQ tab, drag across the Low (L), Middle (M), or High (H) text box.

2 To switch to a different bank of equalization settings, click the Eq/A or Eq/B button. (Double-click the button to copy the current settings to the other bank.)

To equalize a track by using the Track Equalizers window:

1 Select the track, and then choose Window > Track EQ. Alternatively, click the EQ tab in the track controls, and then right-click the H, M, or L box.

2 Set the desired options.

For more information, search for "Track Equalizers options" in Help.

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ug.book Page 184 Tuesday, March 16, 2004 1:29 PM

Working with ReWire tracks

To work with ReWire tracks, you must first set up ReWire connections, assigning ReWire outputs to one or more tracks in a session. (See "Setting up ReWire connections" on page 42.) ReWire tracks offer similar controls to audio tracks. For example, you can quickly change volume, pan, and equalization settings, or you can apply real-time effects. Similarly, you can assign a different device to a ReWire track at any time. Note, however, that saved sessions store only changes made in Adobe Audition; be sure to also save any changes made in the ReWire slave application.

When you synchronize via ReWire, you link the transport controls and timeline of Adobe Audition and the ReWire slave application. For example, if you click the play button in the slave application, Adobe Audition plays the linked session, sending the audio through the outputs specified in the Device Properties dialog box. (See "Setting properties for audio output devices" on page 37.) You can also, however, preview individual modules in the slave application to hear them independently of the Adobe Audition session. When you do, the modules send audio through the sound card specified in the Sounds and Audio Devices control panel.

If you notice a timeline offset between Adobe Audition and the slave application, lower the Playback Buffer Size on the Multitrack tab of the Settings dialog box (choose Options > Settings). The default value is 1, but you can enter values as low as 0.1. Because extremely low buffer sizes may cause audio to drop out, you may need to try different values to find one that is acceptable.

To assign a different device to a ReWire track:

1. In the track controls, click the RW button.
2. For Device Type, select ReWire.
3. Select the device, and click OK.

To convert a ReWire track to an audio track:

Right-click the ReWire track, and choose Bounce.

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ug.book Page 185 Tuesday, March 16, 2004 1:29 PM

Working with MIDI tracks

Audition can import MIDI files as clips on MIDI tracks. MIDI tracks contain a subset of the controls available for audio tracks: a name text box and controls for solo, mute, and volume. MIDI tracks also, however, contain one unique control: a Map button for assigning MIDI output devices.

Adobe Audition doesn't include MIDI clips in exported mixdown files. You can, however, convert MIDI clips to audio clips by recording the output of a MIDI sound module on an audio track.

Note: *In general MIDI terminology, MIDI tracks are instrument tracks in MIDI files. In Adobe Audition, however, MIDI tracks contain MIDI clips, which in turn contain instrument tracks.*

For more information, search for "Working with MIDI" in Help.

Using real-time effects

In Multitrack View, you can apply real-time effects to audio and ReWire tracks. With these flexible effects, you can adjust effects settings as a mix plays. Because real-time effects are nondestructive, you can remove them from a track at any time. You can also change the order of effects to produce a different sonic texture. (For example, you can place Reverb prior to Sweeping Phaser, or vice versa.)

To change an effects mix over time, use clip envelopes. (See "Automating mixes with clip envelopes" on page 188.)

Applying and removing real-time effects

You can apply real-time effects by using either the Organizer window or the Effects Rack dialog box. To remove or reorder these effects, however, you must use the Effects Rack dialog box. You can also use this dialog box to save groups of real-time effects as a preset, which you can quickly apply to multiple tracks.

Clicking the FX button to access the Effects Rack dialog box

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ug.book Page 186 Tuesday, March 16, 2004 1:29 PM

To apply a real-time effect to a track:

From the Effects tab of the Organizer window, drag the effect to the track.

If the Organizer window is closed, right-click the FX button in the track controls, choose Rack Setup, and add the effect in the Effects Rack dialog box.

To change settings for a previously applied effect:

1. In the track controls, right-click FX, and choose FX Settings.
2. Click the tab for the effect, and change settings as desired.

To remove or reorder a real-time effect:

1. In the track controls, right-click FX, and choose Rack Setup.
2. Select the effect in the Current Effects Rack list, and then do one of the following:

- To remove the effect, click Remove.
- To reorder the effect, click either Move Up or Move Down.

To create or apply an effects group preset:

1. In the track controls, right-click FX, and choose Rack Setup.
2. Do one of the following:

- To create a preset, click New, and type a name for the preset.
- To apply a preset, choose it from the Preset menu, and click Apply.

Mixing real-time effects

In the FX mixer, you can change the ratio of dry to wet sound, bypass effects, and combine effects as serial or parallel groups. By default, multiple effects are combined in serial groups, in which the signal travels directly from the output of one effect to the input of the next. In parallel groups, each effect independently receives the dry signal, and the effect outputs are mixed at equal levels.

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ug.book Page 187 Tuesday, March 16, 2004 1:29 PM

When you click Serial or Parallel in the FX mixer, mix settings automatically change to achieve the results above. For serial groups, effect inputs are set to 0% of the dry source (specified in the Src text box) and 100% of the previous effect (specified in the Prv text box). Likewise, all effect output sliders are set to zero except for the final slider, which is set to 100%. For parallel groups, effect inputs are set to 100% of the dry source and 0% of the previous effect, while effect output sliders are set to an equal level (33% each for three effects, 25% each for four effects, and so on).

Note: *The first effect in the FX mixer lacks Src and Prv text boxes because no previous effect exists.*

FX mixer

To mix real-time effects:

1 In the track controls, right-click the FX button, and choose FX Mixer.

2 Do any of the following:

- To change the ratio of dry to wet sound that the track outputs, move the Dry Out slider and the effects sliders.
- To bypass an effect, click Bypass.
- To combine effects in serial or parallel groups, click either Serial or Parallel.

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ug.book Page 188 Tuesday, March 16, 2004 1:29 PM

- To change the ratio of dry to wet audio that an effect receives, enter percentages in the Src and Prv text boxes. (Src represents the dry sound; Prv represents the output of the previous effect.)

To bypass all real-time effects for a track, right-click the FX button in the track controls, and choose Bypass.

Locking tracks with real-time effects

After you apply real-time effects to a track and edit them, you can lock the track to save processing power for other mixing tasks—an important consideration for complex mixes. Adobe Audition stores locked tracks in the background mix, removing them from the CPU load.

If a track is locked, you can't edit effects, clips, or envelopes it contains. You can quickly unlock the track, however, if you need to change it. Though locking tracks takes a small amount of processing time, unlocking tracks is instantaneous.

To lock or unlock a track that has real-time effects:

In the track controls, click Lock.

Automating mixes with clip envelopes

With clip envelopes, you can automate volume, pan, and effects settings over time. For example, you can automatically increase clip volume during a critical musical passage and later reduce the volume in a gradual fade out. For tracks with real-time effects, you can also automatically change the ratio of dry to wet sound.

Envelopes operate nondestructively, so they don't change the original audio file in any way. If you open an original file in Edit View, for example, you won't hear the effect of any clip envelopes. Envelopes also operate in real-time, so you can edit them as a mix plays.

You can identify envelopes by color and initial position. For example, volume envelopes are green lines initially placed across the top of clips. Pan envelopes are blue lines placed in the center of clips. You edit envelopes by dragging control points on these lines. With volume envelopes, for example, the top of a clip represents 100% of track volume, while the bottom of a clip represents full attenuation (silence). With pan envelopes, the top of a clip represents full left, while the bottom represents full right. If an envelope is too high or low, preventing you from raising or lowering control points, you can rescale it.

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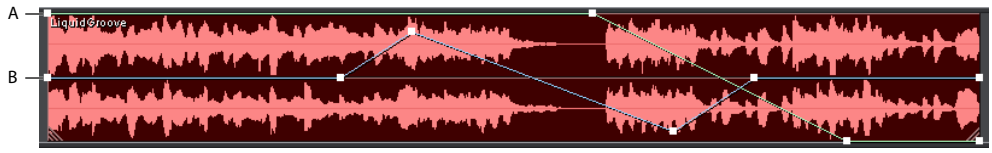
ug.book Page 189 Tuesday, March 16, 2004 1:29 PM

Note: *Wet/dry mix envelopes have the same initial position as volume envelopes, so you may need to hide one to reveal the other.*



Two envelopes in the track display
A. *Volume envelope* ***B.*** *Pan envelope*

To show or hide envelopes:

In the toolbar, click any of the following buttons:

- Show Volume Envelopes .
- Show Pan Envelopes .
- Show Wet/Dry Mix Envelopes .
- Show FX Parameter Envelopes .
- Show Tempo Envelopes .

You cannot edit tempo envelopes, which display the tempo of MIDI clips.

To edit a clip envelope:

1. In the toolbar, click the Edit Envelopes button .
2. Select the clip, and then do any of the following:

- To add a control point, click the envelope.
- To remove a control point, drag it off the clip.
- To move a control point, drag it. (To maintain time position, hold down Shift while dragging.)
- To move all control points up or down by the same percentage, hold down Ctrl while you drag.
- To move all control points up or down by the same amount, hold down Alt while you drag. (This option retains envelope shape, restricting movement to the limits defined by the highest and lowest control points.)

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ug.book Page 190 Tuesday, March 16, 2004 1:29 PM

Note: *With MIDI clips, volume envelopes control MIDI velocity, which usually represents the force with which a note is struck. However, some synthesizers are programmed so that velocity changes pitch or harmonic content.*

To clear all control points for an envelope:

Right-click the clip containing the envelope, and choose Envelopes > [envelope type] > Clear Selected Points.

To use spline curves for an envelope:

Right-click the clip containing the envelope, and choose Envelopes > [envelope type] > Use Splines.

To rescale a volume envelope:

1 Right-click the clip containing the envelope, and choose Rescale Volume Envelopes.

2 Enter the number of decibels by which you want to raise or lower the envelope. Possible values range from –40 to 40. Negative values raise envelopes and lower clip volume by an equal amount; positive values do the opposite.

You can also rescale all volume envelopes in a session.

Using the Mixers window

The Mixers window consists of the Track Mixer and Bus Mixer tabs, as well as a slider that controls the master volume of your session.

The Track Mixer tab mimics a real-world mixing console. It gives you an alternative view of a session, providing a broader overview than the track display, especially if you're working with more than a handful of tracks at once.

The Bus Mixer tab lets you create, configure, and control up to 26 buses. With buses, you can group related tracks and collectively adjust volume or apply real-time effects.

Even if you don't use the Mixers window, consider docking and resizing it so that only the master volume slider is visible. You can use this slider to quickly optimize the overall volume of a mix.

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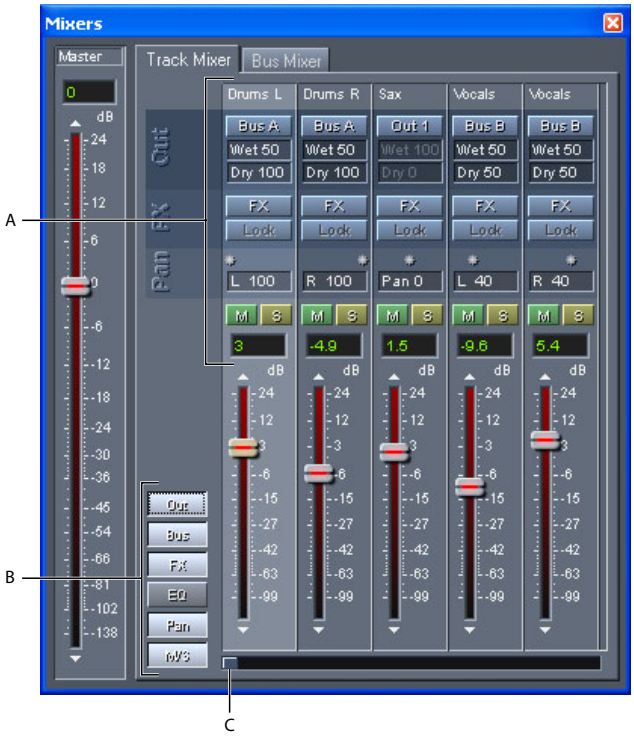
ug.book Page 191 Tuesday, March 16, 2004 1:29 PM

Using the Track Mixer

The Track Mixer provides another method for viewing the tracks in a session. Though it lacks the waveforms, clips, and envelopes visible in the track display, the Track Mixer lets you view and edit more tracks simultaneously.

To automate volume and pan changes over time, use clip envelopes. (See "Automating mixes with clip envelopes" on page 188.)



The Track Mixer
***A.** Track controls **B.** Control display buttons **C.** Scroll bar*

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ug.book Page 192 Tuesday, March 16, 2004 1:29 PM

To use the Track Mixer:

1 In Multitrack View, choose Window > Mixer.

2 Click the Track Mixer tab, and set the following options:

Control display buttons Lets you customize the appearance of the Tracks Mixer. Each of the six buttons—Out, Bus, FX, EQ, Pan, and M/S—displays a different track control.

- Out shows and hides the Out buttons.
- Bus shows and hides the Wet and Dry text boxes for buses.
- FX shows and hides the FX and Lock buttons.
- EQ shows and hides the three equalization text boxes (H, M, L).
- Pan shows and hides the Pan controls.
- M/S: Shows and hides the Mute and Solo buttons.

Out 1 Opens the Adobe Audition Playback Devices window, which lets you assign the output properties for the selected track. The button's label changes to reflect the output device (for example, Device 2 or Bus C).

Wet and Dry text boxes Control the ratio of processed to original signal that tracks output to buses. To change these values, either enter a value or drag across the text boxes. (You can also change these values on the Bus tab of the track controls.)

To output a track to a bus, see "Specifying track input and output devices" on page 181.

FX Opens either the Effects Rack (if the track doesn't yet have an effect assigned to it, regardless of what effects may be assigned if the track is part of a bus) or the dialog box for whatever effect is assigned to the track.

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ug.book Page 193 Tuesday, March 16, 2004 1:29 PM

Lock Locks or unlocks a track. If the Lock button is disabled, the track has no effects.

H, M, L Show the amplitude of high, mid, and low equalization frequencies applied to the track. To change one of these values, drag across the text box. Dragging to the right increases the value, while dragging to the left reduces it.

Pan controls Adjust the balance of each track. To use the control that looks like an asterisk, drag it to one of three positions: hard left, zero pan, and hard right. The Pan text box provides a more precise way of adjusting pan. To change a pan value, drag across the text box to the left or right.

Mute and Solo buttons Let you mute or solo a track. Click the Mute button for as many tracks as you like to turn off their output. Click the Solo button to solo the track. To solo multiple tracks, hold down the Ctrl key as you click the Solo buttons.

Track faders Adjust the track's relative volume in the mix. Move the slider up (or click the triangle above it) to increase the volume; move it down (or click the triangle below it) to reduce the volume. Alternatively, enter a value (in decibels) in the text box above the slider.

Scroll bar Lets you scroll from tracks 1 to 128 and any point in between.

Using the Bus Mixer

Adobe Audition gives you the ability to organize multiple tracks into buses, which are especially useful for grouping related tracks and collectively adding real-time effects or adjusting volume. For example, you can output four tracks of background vocals to one bus, and then apply one reverb effect to that bus. (Individually applying the same reverb to each vocal track would inefficiently drain CPU resources.) You can create up to 26 buses.

After you create and configure a bus, you can output tracks to it. (See "Specifying track input and output devices" on page 181.) Then, on the Bus tab of the track controls, you can change the ratio of wet to dry sound that tracks send to assigned buses.

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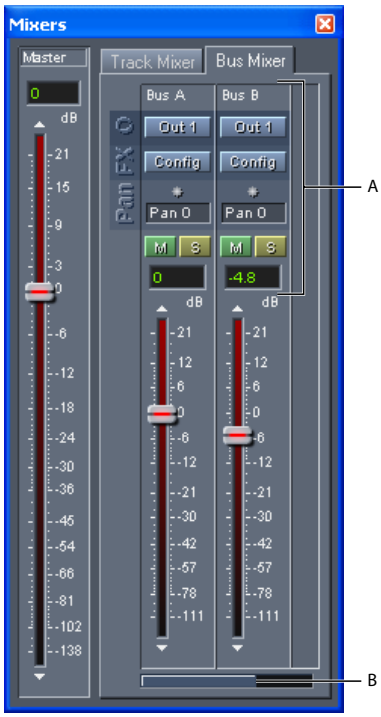
ug.book Page 194 Tuesday, March 16, 2004 1:29 PM



The Bus Mixer
***A.** Bus controls **B.** Scroll bar*

To create and configure a new bus:

1 In Multitrack View, choose Window > Mixer.

2 Click the Bus Mixer tab, and then click New in the right-most Bus channel.

3 In the Bus Properties dialog box, enter a name in the Friendly Name field, and select an output device.

4 In Installed Real-Time Effects list, select effects for the bus, and click Add.

5 Click OK.

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ug.book Page 195 Tuesday, March 16, 2004 1:29 PM

To mix and reconfigure buses:

1 In Multitrack View, choose Window > Mixer.

2 Click the Bus Mixer tab, and then set the following options:

Out Opens the Bus Properties dialog box, where you can specify a different output device or combination of effects.

Config Opens the configuration window for the selected bus. Here you can access the parameters for each effect added to the bus. Sliders also let you adjust the volume of all effects in the bus, as well as the desired Dry Out level.

- Click Serial to connect the effects on the bus in sequence, with the output of one effect connected to the input of the next.
- Click Parallel to connect the effects on the bus separately, mixing only their outputs together.
- Click Rack Setup to open the Properties dialog box for the bus.

Pan controls Adjust the balance of each track. To use the control that looks like an asterisk, drag it to one of three positions: hard left, zero pan, and hard right. The Pan text box provides a more precise way of adjusting pan. To change a pan value, drag across the text box to the left or right.

Mute and Solo buttons Let you mute or solo a bus. Click the Mute button for as many buses as you like to turn off their output. Click the Solo button to solo the bus. To solo multiple buses, hold down the Ctrl key as you click the Solo buttons.

Bus faders Adjusts the relative volume of the bus in the mix. Move the slider up (or click the triangle above it) to increase the volume; move it down (or click the triangle below it) to reduce volume. Alternatively, enter a value in decibels in the text box above the slider.

Scroll bar Lets you scroll through buses if all of them don't fit in the Bus Mixer tab.

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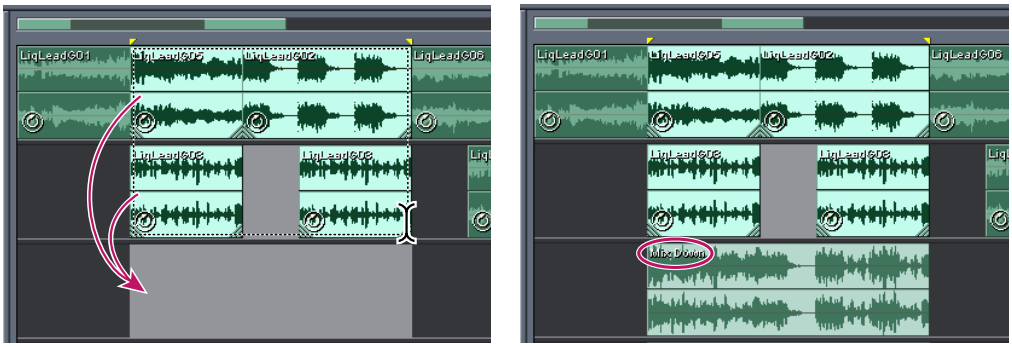
ug.book Page 196 Tuesday, March 16, 2004 1:29 PM

Mixing down ReWire tracks and specific audio clips

You can mix down ReWire tracks and specific audio clips to a new file that opens in Edit View, an empty track in the current session, or a track in a CD project.

You can also mix down entire sessions, exporting them in a variety of formats. (See "Saving and exporting sessions" on page 228.)



Mixing down specific audio clips to an empty track

To mix down ReWire tracks and specific audio clips:

1 Select any audio clips you want to mix down.

2 Select the range you want to mix down.

If you want to mix down complete clips, you can skip step 2 if the session doesn't contain ReWire tracks.

3 From the Edit menu, choose Mix Down To File, Mix Down To Empty Track, or Mix Down To CD Project, and then choose one of the following:

- All Audio Clips to mix down ReWire tracks and all audio clips.
- Selected Audio Clips to mix down ReWire tracks and selected audio clips.
- All Audio Clips (Mono) to mix down ReWire tracks and all audio clips in mono.
- Selected Audio Clips (Mono) to mix down ReWire tracks and selected audio clips in mono.

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ug.book Page 197 Tuesday, March 16, 2004 1:29 PM

Chapter 8: Using Loops

Loops let you use one sound file in a variety of compositions, each with a different tempo and key.

About loops

Loop-based song creation has recently cropped up in nearly all musical circles. From best-selling pop, rap, and hip hop songs to the alternative, adult contemporary and jazz realms, using loops, even as basic rhythm tracks, is a very appealing and modern technique for making music. With Adobe Audition, you can create your own loops or access any of the thousands supplied in the Adobe Audition Loop Library.

Loops typically contain one to two bars of music. Most pop and rock music follows a 4/4 time signature, meaning that one bar has four beats, two bars have eight beats, and so on.

With loops in Adobe Audition, you can do the following:

- Change the pitch and timing of loops independently of each other, so you can easily incorporate the same loop into many different Adobe Audition sessions and musical compositions.
- Quickly and easily add or subtract repetitions of a loop by dragging with the mouse. (With snapping enabled, this method applies even to individual beats within a loop. For example, you can drag to create 1.5 repetitions and end precisely on a snare hit at a loop's midpoint.)
- Snap other audio clips to loop end-points and beats within the loop.

Working with loops is typically a three-step process, in which you select part of a waveform, specify its properties in Edit View, and then use the resulting loop in compositions in Multitrack View.

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ug.book Page 198 Tuesday, March 16, 2004 1:29 PM

Defining loops

To come up with a good loop, you first need to select and save a waveform that, when played over and over, repeats precisely on a beat. This process is called *defining* a loop. Although defining on a beat isn't absolutely required, doing so makes loops more useful because you can combine them in rhythm with other loops.

Selecting a waveform that starts and ends on a clear beat helps make for a good loop.

To define a loop:

1 Open the waveform from which you want to define the loop.

2 Switch to Edit View.

3 Choose Edit > Auto-Cue > Find Beats And Mark. Set up the dialog box to find the beats in the waveform, and click OK. See "Setting cues automatically" on page 101 for more information on the Find Beats And Mark command, and see "Finding beats" on page 90 for information on setting up Adobe Audition to find beats.

The beats in the waveform are now indicated, helping you select a start and end that lands on the beat.

4 Choose Edit > Snapping > Snap To Zero Crossings. (See "Snapping" on page 91.).

This step makes your selection snap to places in the waveform that have zero amplitude, preventing audible noise at the beginning and end of the loop.

5 Select the part of the waveform you want to define, typically starting and ending on a beat.

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ug.book Page 199 Tuesday, March 16, 2004 1:29 PM

6 Click the Play Looped button ∞ to repeatedly play your selection.

7 Adjust the start and end of the selection until just the material you want is selected.

8 Choose View > Display Time Format > Edit Tempo to specify detailed tempo information for the loop. (See "Calculating the tempo of selected ranges" on page 199.)

9 Choose Edit > Copy To New. This step copies the selected area to a new file where you can set the loop's properties. (See "Setting permanent loop properties in Edit View" on page 200.)

Calculating the tempo of selected ranges

In both Edit View and Multitrack View, you can calculate the tempo of a selected range by using the Edit Tempo command. This command lets you quickly determine loop tempo in Edit View or change session tempo in Multitrack View. It also lets you change the beats-per-minute (bpm) value for horizontal rulers in the Bars and Beats time format.

To calculate the tempo of a selected range:

1 Choose View > Display Time Format > Edit Tempo.

2 Set the following options, and click OK:

Beats Highlighted/Bars Highlighted Specifies the number of beats or bars highlighted in the selection according to the Bars and Beats format. This number will probably be wrong initially, because you haven't defined the tempo yet. In this case, enter the correct number of bars to use for extracting tempo information.

Extract Calculates tempo information from the highlighted selection, and fills in the Beats per Minute and Offset values. Before clicking Extract, make sure to enter a value for Beats per Bar.

Current Beat At Defines the bar and beat information for the selection's starting point (or the current cursor position if no selection has been made). Adobe Audition assumes that this represents a downbeat. Changing this value updates the Song Start value based on the current tempo settings.

Reset 1:1 To Cursor Changes the Current Beat At value to 1:1.00.

Song Start Represents the number of milliseconds before the measure 1:1.00 begins. This value is for information only.

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ug.book Page 200 Tuesday, March 16, 2004 1:29 PM

Beats Per Minute Displays the number of beats that occur in a one-minute interval. You can calculate this value by clicking Extract.

Beats Per Bar Assigns the number of beats that make up one measure/bar. For instance, enter **4** for 4/4 time, **6** for 6/8 time, and the like.

Beat Length Specifies the value of the beat. For instance, enter **2** for a half note, **4** for a quarter note, and **8** for a sixteenth note.

Ticks Per Beat Specifies the number of sections each beat is divided into, or the value after the decimal point. You can enter a number between 2 and 3600. For instance, if you enter 32 ticks per beat, then a time setting of 4:2:16 represents an eighth note (a note halfway) between beats 2 and 3 in 4/4 time.

Setting permanent loop properties in Edit View

After you define a loop, you can set permanent loop properties so it works well with other clips in a session. The Loop Info tab of the Wave Properties dialog box lets you specify these properties, such as number of beats, default tempo, and musical key. Setting permanent loop properties makes a loop far easier to work with in Multitrack View.

Loop properties that you set in Edit View are saved with the file and are permanent. Loop properties that you set in Multitrack View are saved with the session and aren't permanent. In addition, session-based loop properties in Multitrack View override Edit View loop properties. See "Setting impermanent loop properties in Multitrack View" on page 202 for more information.

To set loop properties in Edit View:

1. Choose View > Wave Properties.
2. Click the Loop Info tab.
3. Set any of the following options, click OK, and then save the file:

Loop Tells Adobe Audition that the file is a loop. If the file is inserted into a Multitrack session, looping is enabled automatically for that audio clip.

One Shot Indicates that the file plays once rather than repeats like a loop.

Number Of Beats Specifies the number of beats in the loop. Adobe Audition attempts to detect and specify the number of beats for you, but you can adjust the value if necessary.

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ug.book Page 201 Tuesday, March 16, 2004 1:29 PM

Tempo Specifies the number of beats per minute in the loop. Adobe Audition calculates this value automatically based on Number Of Beats. Don't worry if the value isn't a whole number—for example, 80.4 instead of 80—after you loop the file, Adobe Audition can stretch it to whatever tempo you want.

Key Specifies the loop's key, so that if you create a session and want to adjust the key of all audio clips globally, Adobe Audition has a reference for each file. If a loop file is a drum track, choose Non-Voiced. This option is especially important if you plan to change the key of multiple loops in a session, because you won't want to pitch-shift a drum track to the key of E (for example) if it has no key to begin with.

Find Nearest Scans the loop to locate the nearest key. This option works best with monophonic files (that is, solo instruments). Because many keys share the same notes in the scale, you can think of this setting as root note for transposition.

Stretch Method Specifies how (if at all) the loop stretches to match the session's tempo. Choose one of the following settings:

- Fixed Length (No Stretching) causes the loop to play at its native tempo no matter what the session tempo is set to. If a session has multiple loops of different tempos, and each is set to Fixed Length, no two loops will seamlessly match in tempo. This setting is useful if you plan to insert and loop a file in a session where you don't plan on doing any type of time stretching or pitch shifting. The most common uses for this setting is inserting a pattern over live music or using one to underscore live vocals.
- Time-Scale Stretch stretches the file (just like the Stretch effect) to match the tempo of the session. Corresponding options are Quality (High, Medium, or Low), Frame Size (the number of splices per beat), and the percentage of Frame Overlapping. This method stretches a file based on its actual length, so you should use it if you loop something like a synth pad or a sustained string section (which don't have actual beats, per se). This method is most commonly used for "tonal" instruments, like piano, bass, and guitar.
- Resample (Affects Pitch) resamples the loop to match the session's tempo, affecting the pitch. High, Medium, and Low Quality options are available. This method is commonly used in R&B and hip hop tracks, primarily because you can achieve exaggerated stretching and compressing of files. If loops set to Resample are time stretched, their pitch changes. This setting is most commonly used on drum tracks to create a lo-fi, dirty, phat kind of sound. It can also work well on voice and voiceovers if you're trying to change the sound and timbre of a speaker's voice.

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ug.book Page 202 Tuesday, March 16, 2004 1:29 PM

- Beat Splice loops the file based on beats detected in it, similarly to the Find Beats And Mark command. (See "Defining loops" on page 198.) This setting works only on loops that have very sharp and short sounds, like drum tracks. If the waveform already has beat markers, you can select Use File's Beat Marks to use them. Otherwise, Auto-Find Beats is selected. If necessary, you can change the corresponding default values of 10 dB and 9 milliseconds to find the beat.
- Hybrid uses the current Time-Scale Stretch settings if you lower the bpm (beats per minute), and it uses the current Beat Splice settings if you raise the bpm.

Setting impermanent loop properties in Multitrack View

Loop properties that you set in Multitrack View are saved with the session and aren't permanent, but they override any permanent loop properties you've set in Edit View. (See "Setting permanent loop properties in Edit View" on page 200.)

By default, changes made to a looped audio clip in Multitrack View affect only that clip, unless Adjust All Loop-Enabled Clips That Use This Wave is selected in the Loop Properties dialog box.

To set impermanent loop properties in Multitrack View:

1. Select an audio clip.
2. Choose Edit > Loop Properties.
3. In the Audio Clip Looping dialog box, set the following options, and click OK:

Enable Looping Sets the file so that you can loop the audio clip by dragging its right edge.

Simple Looping (No Gaps) Makes the audio clip loop continuously, with no spaces between looping instances.

Repeat Every X Seconds Repeats the loop at the number of seconds you specify. If loop information is already entered for the audio clip, proper values for Repeat Every X Seconds and Repeat Every X Beats are entered automatically so that the audio clip loops continuously at the proper tempo. If you change the Repeat Every X Seconds value, Adobe Audition ignores the tempo and stretches the file to finish its loop in the specified number of seconds. Normally, you should select this option and enter the number of beats in the Source Waveform Information area.

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ug.book Page 203 Tuesday, March 16, 2004 1:29 PM

Repeat Every X Beats Repeats the loop at the number of beats you specify. If loop information is already entered for the audio clip, proper values for Repeat Every X Seconds and Repeat Every X Beats are entered automatically so that the audio clip loops continuously at the proper tempo. If you change the Repeat Every X Beats value, Adobe Audition stretches the file to finish its loop in the specified number of beats. However, you'll generally want to select Repeat Every X Beats and enter the number of beats in the Source Waveform Information area.

Follow Session Tempo Plays the loop at the session's tempo instead of its native tempo. For example, if you play a 100 bpm loop in a 120 bpm session, the loop is stretched to 120 bpm. Selecting this option disables the BPM text box. If you don't select this option, the loop plays at the tempo specified in the BPM text box.

Lock Position To Tempo Locks the left edge of the audio clip to the bar/beat. If you change tempo, the audio clip moves so that it starts at the same beat. Normally, you should select this option if you stretch to tempo. In addition, you can select this option for a one-shot clip that's not a loop (like a thunder clap or a gong) if you want it to start in time with other music that's aligned to the session's tempo.

Source Waveform Information Specifies settings for the source waveform. (See "Setting permanent loop properties in Edit View" on page 200.)

Tempo Matching Specifies settings for matching the tempo of the loop to the rest of the sound file you're working with. (See "Setting permanent loop properties in Edit View" on page 200.)

Transpose Pitch Transposes the pitch of the looped clip by the specified number of half-steps. Positive numbers raise the pitch, and negative numbers lower it.

Adjust All Loop-enabled Clips That Use This Wave Globally changes the settings for all clips that reference the same waveform. For example, if you insert the same loop file into Multitrack View four times, and you then adjust the loop properties on one of its clips, the other three instances of the loop in the session are adjusted, too.

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ug.book Page 204 Tuesday, March 16, 2004 1:29 PM

Setting the tempo, time signature, and key for sessions

The Session Properties window lets you specify the tempo, time signature, and key for loops in a session. All loop-enabled clips automatically adjust to match new settings; regular clips are unaffected.

To preview loop files at the tempo and key of a session, select either the Loop option in the Insert Audio dialog box, or the Follow Session option in the Files tab of the Organizer window. (See "Inserting audio files into multitrack sessions" on page 63 and "Previewing audio by using the Organizer window" on page 77.)

To set the tempo, time signature, and key for a session:

1 In Multitrack View, choose Window > Session Properties if the window isn't visible.

2 Set any of the following options:

Tempo Specifies the tempo of the session, measured in beats per minute.

Beats/Bar Specifies the number of beats per bar.

Key Specifies the session's key.

Time Specifies the session's time signature. Choosing a different time signature automatically updates the Beats/Bar setting.

Advanced Opens the Advanced Session Properties dialog box so that you can set advanced properties for a session, such as a time offset, a customized metronome, and notes about the session. (See "Setting advanced session properties" on page 167.)

Metronome Toggles the built-in metronome on and off. (See "Setting advanced session properties" on page 167.)

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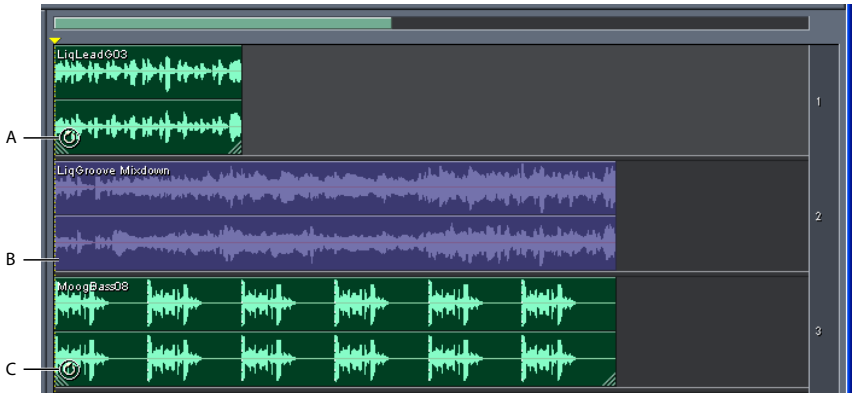
Working with loops in the track display

After you add loops to a multitrack session, you can edit them in the track display, extending them to repeat as needed, and synchronizing them with the beat of the music.



Loops in the track display
***A.** Single loop **B.** No loop **C.** Extended (repeated) loops. Even though loop files are short, you can extend them to repeat as many times as needed.*

To synchronize loops to musical beats:

1 Choose View > Display Time Format > Bars And Beats to change the format of the ruler to bars: beats:ticks per beat. This format makes it easier to visually line up loops with musical beats. (See "Monitoring time" on page 69.)

2 From the Edit > Snapping submenu, choose any of the following

- Snap To Ruler (Coarse) to snap loops to the beats within bars. Use this option if you work with 1/4 or 1/2 bar loop files. (See "Snapping" on page 91.)
- Snap To Clips to snap loops to the start or end of audio clips. (See "Snapping clips to loop endpoints and other clips" on page 170.)
- Snap To Loop Endpoints to snap loops to the start or end of other loops. (See "Snapping clips to loop endpoints and other clips" on page 170.)

Also consider snapping non-loop-enabled audio clips to the beat and each other, so that all clips are aligned. You can snap the current-time indicator, too.

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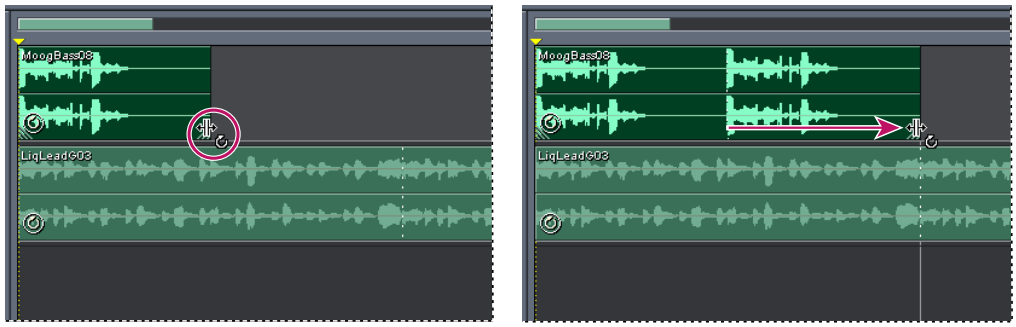
ug.book Page 206 Tuesday, March 16, 2004 1:29 PM

To extend or shorten a loop-enabled clip:

1 Select the clip, and then position the pointer over the bottom left or right handle—the loop editing icon appears.

2 Drag the handle to extend the loop the desired number of bars. Depending on how far you drag, you can make the loop repeat fully or partially. For example, you might drag a loop that is one bar long so that it extends 3-1/2 bars, ending on a beat within the loop. As you cross each bar, a white vertical line appears in the clip. This is the snap-to line, indicating perfect alignment to beats in other tracks.



A B

Extending a loop
A. *Moving the cursor to activate the loop.* ***B.*** *Dragging the loop, with snap to lines indicating how the loop snaps to the beats in other tracks*

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Chapter 9: Working with Video

Adobe Audition tightly integrates with digital video, enhancing any video project with professional sound. To create uniquely sophisticated soundtracks, you can combine Adobe Audition with Adobe Premiere Pro® and Adobe After Effects® to take full advantage of Adobe Audition's flexible mixing features.

About working with video

With Adobe Audition, you can improve the sound of any video project. For example, if you need to improve the audio quality of an existing soundtrack, you can use Edit View to quickly restore and enhance the audio. Or, if you want to create elaborate soundtracks with flexible, real-time mixing tools, you can use Multitrack View to preview video, add audio and MIDI tracks, and export entirely new soundtracks. (See "Importing audio and video from video files" on page 208.)

For maximum flexibility, you can combine Adobe Audition with Adobe Premiere Pro and After Effects. Tight integration between these products lets you quickly remix a soundtrack as the needs of a video project change over time.

Working with Adobe Premiere Pro and After Effects

If you use Adobe Premiere Pro or After Effects, you can easily remix and edit soundtracks in Adobe Audition. To do so, first configure Adobe Audition to link session files with exported audio mixdowns in WAV format. Once these files are linked, you can select an imported mixdown file in Adobe Premiere Pro or After Effects, and then remix the related session in Multitrack View, or edit the mixdown file in Edit View.

To link session files with exported audio mixdowns in WAV format:

1 Choose Options > Settings, and then click the Data tab.

2 Select Embed Project Link Data For Edit Original Functionality, and then click OK.

3 When you export mixdown files, select Save Extra Non-Audio Information in the Export Audio dialog box.

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To remix or edit a mixdown in an Adobe Premiere Pro or After Effects project:

1. In the Adobe Premiere Pro or After Effects project, select the mixdown file.
2. Choose Edit > Edit Original.
3. Select one of the following, and then click OK:

- Launch The Audition Multitrack Session Which Created This File.
- Insert This File Into Audition's Edit View.

4. Remix the linked session in Multitrack View, or edit the mixdown file in Edit View.
5. Overwrite the original file by doing one of the following:

- In Multitrack View, choose File > Export > Audio, and specify the same name and location as the original file.
- In Edit View, choose File > Save.

Importing audio and video from video files

In both Edit View and Multitrack View, you can import audio data from a video file in AVI, MPEG, or WMV format. This approach is useful for soundtrack editing that doesn't require a video preview, or for readapting soundtracks for audio-only mediums, such as radio or CD.

Only in Multitrack View, however, can you import both audio and video data from a video file. This approach lets you precisely synchronize audio with a video preview. Note, however, that a multitrack session can contain only one video clip at a time.

To import audio data from a video file:

Do one of the following:

- In Edit View, choose File > Open Audio From Video.
- In Multitrack View, select a track, position the current-time indicator at the desired insert point, and then choose Insert > Audio From Video.

To import audio and video data:

In Multitrack View, select a track, position the current-time indicator at the desired insert point, and then choose Insert > Video.

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ug.book Page 209 Tuesday, March 16, 2004 1:29 PM

Working with video clips

When you import a video file into a multitrack session, video data becomes a video clip on the selected track, and audio data becomes an audio clip on the track below. You can select and move video clips like other clips. (See "Selecting and moving clips" on page 168.) Note, however, that you can also move a video clip independently from the audio clip containing the original soundtrack; to keep related video and audio clips synchronized, group them. (See "Grouping clips" on page 169.)

To synchronize audio and video, you can snap other clips and the current-time indicator to individual frames in a video clip. You can also magnify the session display to view more thumbnails in a clip. These thumbnails serve only as a general guide; for frame-accurate synchronization, use snapping.

Note: *Thumbnails don't appear for MPEG-2 video clips.*

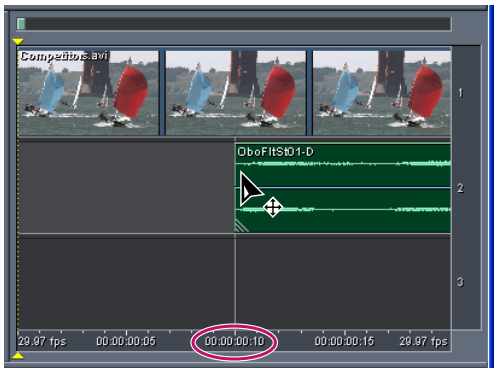
To snap to individual frames in a video clip:

1 Choose View > Display Time Format, and select the SMPTE time format that corresponds to the frame rate of the clip.

2 Choose Edit > Snapping > Snap To Frames.

To view more thumbnails in a video clip:

Horizontally magnify the session display. (See "Zooming" on page 17.)



Snapping to a video frame that falls within a video thumbnail

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ug.book Page 210 Tuesday, March 16, 2004 1:29 PM

Previewing video

In the Video window, you can preview video clips as a multitrack session plays to precisely synchronize a soundtrack with specific video events such as scene changes, title sequences, or special effects. You can customize the preview to optimize it for your monitor size and system speed. For example, you can enlarge the preview to fit a resized Video window or lower the preview quality to increase performance.

The floating Video window in Multitrack View

To hide or show the Video window:

In the toolbar, click the Hide/Show Video Window button .

To automatically show the Video window when you insert a video file:

Right-click the Video window, and select Auto Show Video.

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To customize the video preview:

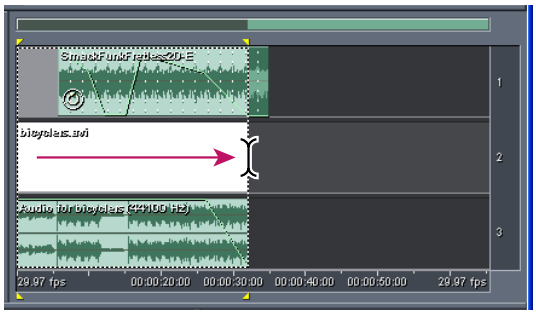
Right-click the Video window, and select any of the following:

- A zoom percentage to zoom in or out.
- Best Fit to fit the preview to the window.
- Maintain Aspect Ratio to maintain that ratio when you resize the window.
- Integer Factor Sizing to constrain the preview to ratios such as 1/2, 1/1, and 2/1 when you resize the window. This option avoids complex resampling, producing a sharper image and increasing performance.
- Low Quality to lower the preview quality.

Note: *Video quality settings take effect when you next import a video clip. To apply a new quality setting to the current clip, close it, and reimport it into the session.*

Preparing video mixdowns for export

In Multitrack View, you can export video mixdowns in AVI format. Video mixdowns combine video clips with audio clips that exist in the same area of the timeline, creating a new soundtrack. Prior to exporting a video mixdown, you can preview it to ensure that it will sound as you expect and, if not, edit the session as desired.



Selecting the start and end points of a video mixdown

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ug.book Page 212 Tuesday, March 16, 2004 1:29 PM

To preview a video mixdown:

1 Choose Edit > Snapping > Snap To Frames.

2 In the session display, select an area that extends from the beginning to the end of the video clip.

3 Play the session, and then do one of the following:

- If the soundtrack doesn't sound as you expect, edit the session as desired, and then repeat steps 2 through 3. (For example, if part of an audio clip is omitted, move the entire clip into the selected area.)
- If the soundtrack sounds as you expect, export a video mixdown. (See "Exporting mixes to video" on page 230.)

You can also use this procedure to export an audio mixdown that you combine with video in a video application, such as Adobe Premiere Pro. Though video mixdowns are limited to stereo audio and AVI format, audio mixdowns support stereo and surround sound in a variety of formats. For more information, see "Exporting mixes to audio" on page 229 and "About surround sound" on page 213.

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ug.book Page 213 Tuesday, March 16, 2004 1:29 PM

Chapter 10: Creating Surround Sound

Adobe Audition includes the Multichannel Encoder, a self-contained dialog box where you can access the tracks of any existing Multitrack session, pan them into the six channels of 5.1 surround sound, and export them.

About surround sound

With surround sound, heard in many popular movies, you can pan an audio mix around the room. Adobe Audition supports 5.1 surround sound, which requires five speakers, plus one low frequency subwoofer (LFE). To properly preview a 5.1 surround-sound mix, your computer must have a sound card with at least six outputs, and the speakers must be connected and positioned as follows:

- Output 1: Front left speaker.
- Output 2: Front right speaker.
- Output 3: Front center speaker.
- Output 4: LFE.
- Output 5: Left surround speaker.
- Output 6: Right surround speaker.

Adobe Audition lets you create and export 5.1 surround sound in a multichannel session by using the Multichannel Encoder dialog box. With this dialog box, you can individually pan each track of a multitrack session to your multichannel setup, preview the current mix, and export the session. You can export your session as six mono WAV files, as one interleaved 6-channel WAV file, or as a Windows Media 9 Pro (WMA) file for use with an external multichannel encoder such as a Dolby or DTS encoder.

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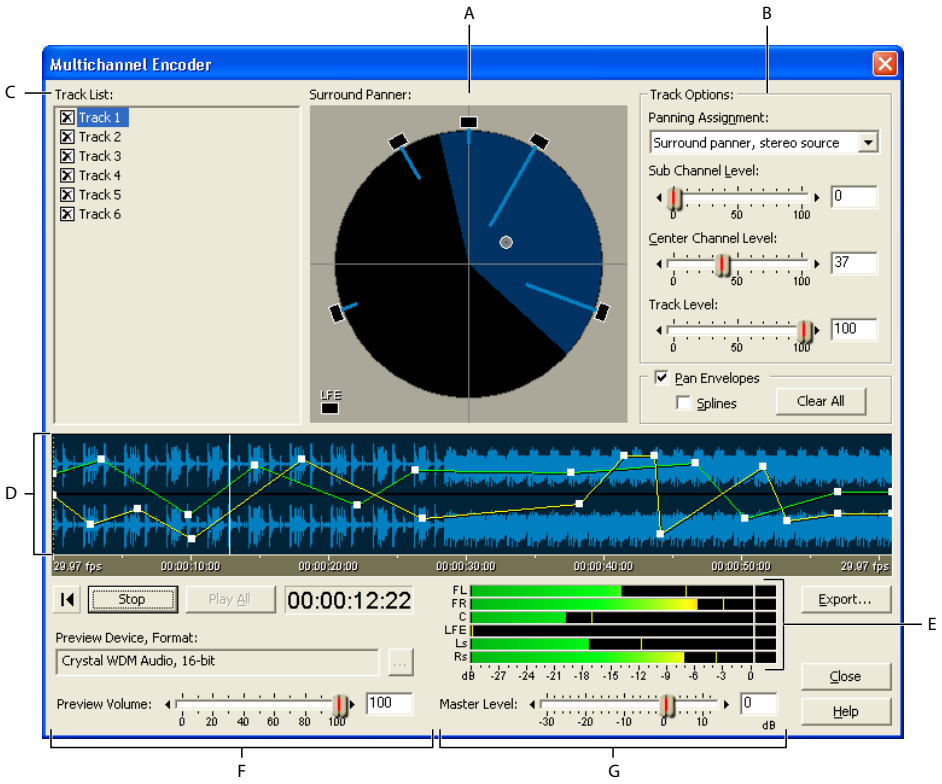
Using the Multichannel Encoder

The Multichannel Encoder dialog box contains several options and controls that let you select tracks and bus outputs, precisely pan audio and adjust volume levels, zoom the waveform display, and preview the project.

To achieve proper 5.1 surround-sound preview playback from the Multichannel Encoder, you need a sound card that offers at least 6-channel analog output, a special interleaved driver that's compatible with the Microsoft DirectSound multichannel format, and Microsoft DirectX 8.0 or later. (Direct X 8.0 is installed by default as part of the Adobe Audition installation; updates are available from the Microsoft Web site.) If your system does not meet these requirements you may receive a warning message and your Play Track and Play All buttons will not be accessible.



The Multichannel Encoder dialog box
***A.** Surround Panner **B.** Track options **C.** Track List **D.** Waveform display with pan envelopes*
***E.** Output meters **F.** Preview controls **G.** Master Volume control*

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ug.book Page 215 Tuesday, March 16, 2004 1:29 PM

To use the Multichannel Encoder:

1 Open an existing Adobe Audition session, or create a new session in the Multitrack window.

2 Once all your tracks are added, achieve a basic stereo mix balance with your desired track volume, stereo pan, and FX settings.

3 Choose View > Multichannel Encoder.

4 In the Track List, select the tracks and bus outputs you want to pan and export. (See "Selecting tracks and buses in the Multichannel Encoder" on page 215.)

5 Under Track Options, specify the Panning Assignment and set the controls as desired. (See "Assigning the panning source" on page 216, "Using the Surround Panner" on page 217, and "Automating the pan envelope" on page 218.)

6 Set the volume for the tracks. (See "Adjusting volume levels" on page 220.)

7 Preview the panned tracks. (See "Previewing the multichannel project" on page 221.)

8 Export the session. (See "Exporting surround-sound files" on page 223.)

Panning tracks and buses for surround sound

Using the Multichannel Encoder, you can pan any of the tracks and buses in your session for surround sound. By panning sound between the six surround-sound speakers, you can make the sound appear to come from anywhere around the listener.

Selecting tracks and buses in the Multichannel Encoder

All of the tracks used in your current multitrack session appear in the Track List in the Multichannel Encoder. If you uncheck a track it is removed from the multichannel preview and is not included when you export the multichannel project.

If you have routed a track to a bus, the bus, instead of the track, will appear in the track list. You can select and pan the bus output as one mono or stereo signal. Additionally, you can access the "dry" track signal and pan it separately as well. (See "Using the Bus Mixer" on page 193 for more information on setting up a bus.)

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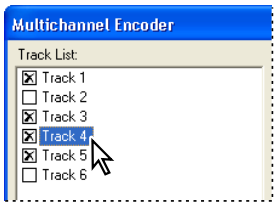
ug.book Page 216 Tuesday, March 16, 2004 1:29 PM

To select a track to pan for surround sound:

In the Multichannel Encoder dialog box, click the name of the track.



The Track List with Track 4 selected

To access a track assigned to a bus in the Track List:

1. Close the Multichannel Encoder.
2. In the track controls in Multitrack View, click the Bus tab for the track you want to access.
3. Increase the Dry value so that it is greater than zero.
4. Reopen the Multitrack Encoder.

Assigning the panning source

At the top right of the dialog box is the Panning Assignment list where you can choose to either use the Surround Panner to position your track sound source or make fixed panning assignments for your track.

To specify the panning assignment:

Choose one of the following options from the Panning Assignment list at the top right of the Multichannel Encoder dialog box:

Surround Panner, Stereo Source Uses the Surround Panner to position your sound source. (See "Using the Surround Panner" on page 217.) It also keeps your stereo left and right signals from your track discrete when panning in the sound field. For example, if your track includes a stereo file, the left stereo signal is sent to the Front Left and Left Surround channels, and your track's right signal is sent to the Front Right and Right Surround channels. The Center channel always receives a summed to mono (L + R) signal. Therefore, as you pan in the five channel sound field, these stereo sources retain their "stereo image" while being routed to the multiple channels.

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Surround Panner, Summed To Mono Lets you use the Surround Panner to position your sound source. However, this option always sums the track's signal to a mono signal. In this mode, panning the sound source to any location in the sound field results in the summed mono signal being fed to all channels.

Lfe Only Sends the entire track signal to the LFE (subwoofer) channel. Your monitoring system applies the proper crossover frequency cutoff for reproducing the audio sent to the LFE channel. Typically, most LFE components in 5.1 surround playback systems are set to a cutoff of < 80 Hz or < 120 Hz. The Multichannel Encoder itself does not apply any filter to the LFE channel audio.

FL + FR, Stereo Sends the selected track's signal as a stereo source directly to only the Front Left and Front Right speakers in a 50/50 stereo balance.

Ls + Rs, Stereo Sends the selected track's signal as a stereo source directly to only the rear Left Surround and Right Surround speakers in a 50/50 stereo balance.

Center + LFE, Stereo When selected for a stereo track, this option routes the track's left channel signal to the Center channel and the track's right channel signal to the LFE (subwoofer) channel discretely. If this option is selected for a track containing a Mono source file, the same signal is sent equally to both the Center and LFE channels. Note that this option is most useful with a stereo source file.

Center Only, Mono; FL Only, Mono; FR Only, Mono; Ls Only, Mono; Rs Only, Mono Sums the selected track's audio to a mono signal, and sends it all to the selected channel. This is the same as dragging the Panner Point directly onto one of the five main speakers in the Surround Panner.

Using the Surround Panner

The Surround Panner is an interactive control representing the audio field. You drag the Panner Point (white dot) to change the perceived sound source. As you move the Panner Point, the light blue Power Indicator lines coming from the speakers change length. The length of the lines indicates the power balance of your sound source coming from each of the five main channels. Additionally, a portion of the sphere appears dark blue to indicate the image of the sound field. That is, when seated in the center of the speakers (marked by the crosshairs), the blue area indicates where the listener perceives the sound to be coming from.

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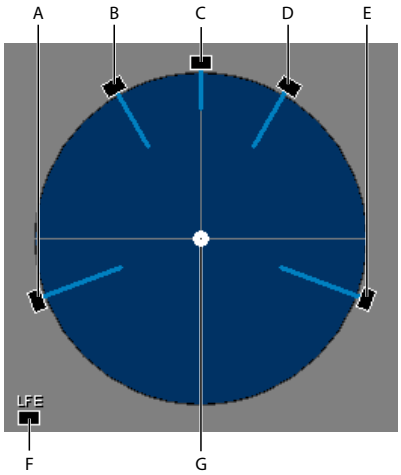
ug.book Page 218 Tuesday, March 16, 2004 1:29 PM

You can also drag the Panner Point outside the sound field directly on top of one of the five main speakers or on top of the LFE speaker. Once the Panner Point is in any one of these speaker locations, the audio from the currently selected track is summed to a mono signal and sent discretely to this one speaker channel. This is an easy way to send the complete track signal all to one channel.



Surround Panner options
***A.** Left Surround **B.** Front Left **C.** Center*
***D.** Front Right **E.** Right Surround*
***F.** Low Frequency Effects (Sub Bass) **G.** Panner Point*

To use the Surround Panner:

In the Multichannel Encoder dialog box, drag the white Panner Point, which represents the location of the audio track in the sound field.

Automating the pan envelope

When you select Pan Envelopes, two envelope lines appear in the waveform display. The yellow envelope line controls the Left/Right balance and the green line controls the Front/Surround balance. These envelopes are interactive with the positioning of the Panner Point in the Surround Panner interface. It is possible to create dynamic panning over time by using these envelopes. (See “Automating mixes with clip envelopes” on page 188.)

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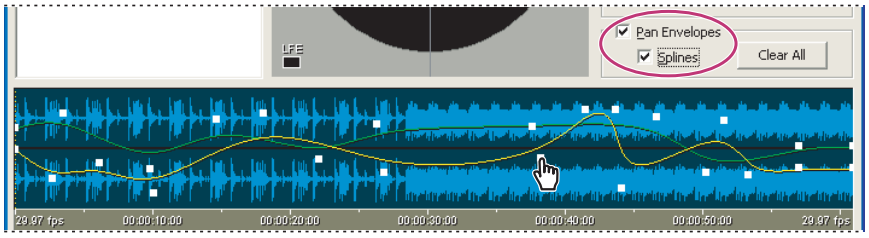
If you prefer to keep your track panned to a fixed point throughout the duration session, deselect Pan Envelopes. Deselecting this option removes the envelopes from the waveform display and lets you set the Panner Point to any static position you want. You can toggle the Pan Envelopes setting on and off and any envelope points you have created for this track are retained. Note that if Pan Envelopes is not selected, you can drag the Panner Point during playback and hear your static pan positioning in real time.



Pan Envelope Automation

To create a dynamic pan on a track:

1 In the Multichannel Encoder dialog box, select the box for a track in the Track List.

2 From the Panning Assignment menu, choose either "Surround Panner, Stereo Source" or "Surround Panner, Summed To Mono".

3 Select Pan Envelopes, located above the right side of the waveform display. Two envelope lines appear in the waveform display. (Because the yellow line starts on top of the green line, you may see only the yellow line until you change the pan position.)

4 Click in the waveform display where you want to set a pan destination for the sound source. The vertical cursor moves to this time location.

5 Drag the Panner Point in the Surround Panner to the desired position in the sound field. Two handle points appear on the envelope lines within the waveform display and move as you position the Panner Point. (You can also click either of the envelope lines to create additional adjustable handles for shaping the envelope lines.)

6 To edit an envelope handle, drag it. The Panner Point moves in tandem to show you the relative position in the sound field during playback. To delete a handle, drag it up or down beyond the edge of the waveform display area.

7 To clear all envelope handle points and reset the track to flat envelopes, select Clear All, located below and to the right of the Pan Envelopes option.

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8 To use spline curves for smoother transitions between points, select Splines.

9 Drag the playback cursor back to the start of the track, and select one of the Play buttons. Watch the Panner Point position, and listen for the dynamic pan setting you just created.

Adjusting volume levels

The Multichannel Encoder lets you adjust the subchannel level, center channel level, and track level.

To adjust the level:

Use any of the following options in the Multichannel Encoder dialog box:

Sub Channel Level Specifies the amplitude of the subchannel level to additionally send the track's signal to the LFE channel.If the currently selected track is assigned to only the LFE channel, this option attenuates the amount of this track's output sent to the LFE channel.

Note: *The Multichannel Encoder does not apply filtering to audio sent to the LFE channel, nor during preview, exporting, or encoding. Therefore, any low-pass filtering needed for your final LFE channel content should be applied to your audio within the Adobe Audition Multitrack, or on your exported .wav files.*

Use a Bass Management circuit in your monitoring setup to ensure that you hear the representative mix levels that might be reproduced in an end listener's playback system.

Center Channel Level Determines the balance of the Front Left, Center, and Front Right channels when in the Surround Panner modes. When set to 100, the Center channel receives an equal percentage of signal as the Front Left and Front Right channels. The position of the Panner Point then determines the positional panning according to this Front Left, Center, Front Right balance ratio.

Track Level Controls the amplitude level of the currently selected track within the Multichannel mix in any selected Surround Panner mode.

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ug.book Page 221 Tuesday, March 16, 2004 1:29 PM

Zooming into and out of the waveform display

There are several options for zooming in and out within the waveform display.

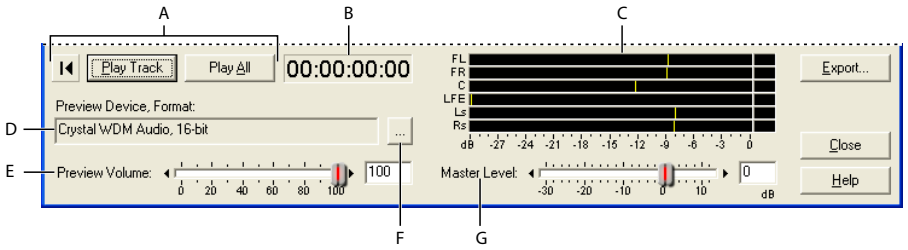
To zoom the waveform:

In the Multichannel Encoder dialog box, do any of the following:

- Place the mouse cursor over the time ruler that runs across the bottom of the waveform display, right-click and select a zoom option.
- Right-drag the desired zoom area on the time ruler. (To zoom back out again, right-click and choose Zoom Out or Zoom Full from the context menu.)
- Place the mouse pointer within the waveform display and turn the mouse wheel. This zooms into the time area directly beneath the mouse pointer. Turning the mouse wheel the other direction zooms back out.

Previewing the multichannel project

The Multichannel Encoder provides several preview options, including different types of playback controls and playback format options.



Preview controls
***A.** Transport controls **B.** Time indicator **C.** Output meters **D.** Preview device **E.** Preview volume **F.** Preview device change button **G.** Master Level*

Previewing a track or session in the Multitrack Encoder

You can choose from or adjust the following preview controls:

Go To Beginning Places the cursor at the start of the track.

Play Track Plays the currently selected track from the cursor location. Playback always plays to the end of the track, regardless of the current zoom level.

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ug.book Page 222 Tuesday, March 16, 2004 1:29 PM

Play All Plays from the cursor location, playing the multichannel mix with all tracks that are checked in the Track List. Like Play Track, Playback always plays to the end of the session, regardless of the current zoom level.

Time indicator Located next to Play All, this indicator shows the time in the preview playback.

Preview Volume Controls the volume of the preview playback, without affecting the volume of the exported files. That is, it doesn't change the amplitude of the exported WAV or encoded WMA files that are created from the Multichannel Encoder, nor does it affect the levels measured by the 6-channel Output Meter. Use the Master Level slider for changing these.

Output meters (FL, FR, C, LFE, Ls, Rs) This set of six meters displays the output of each channel during preview. During Play Track, the meters display the output of only the selected track. During Play All, the meters display the output of the complete 5.1 mix. These levels are what the actual levels will be for your exported WAV or WMA files from the session. You can attenuate the overall 6-channel level by using the Master Level slider beneath the meters.

Master Level Sets the audible level of your preview playback. However, it is primarily intended to adjust the amplitude of the exported or encoded files. Use this slider and reference the meters to optimize the overall peak amplitude of the 5.1 channel mix so that none of the channels are clipping.

Preview Device, Format Displays the currently selected device to which Adobe Audition routes its 6-channel output. (See "Setting the preview device and format" on page 222.)

Setting the preview device and format

The Preview Device, Format option lets you specify the device and format of the previewed audio. This option also displays the currently selected bit rate for preview playback. For information about device requirements, see "Using the Multichannel Encoder" on page 214.

To make changes to the device and bit selections:

1. Click the Change button to the right of the Preview Device, Format option.
2. From the Multichannel Output Device menu, specify the preview sound card.

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ug.book Page 223 Tuesday, March 16, 2004 1:29 PM

Note: *Some sound cards that offer 5.1 playback, such as the Creative Labs Audigy, display only one device driver in the list. In this case, this is the device you should select because the sound card's driver will route the six channels of audio to the correct speakers. (If a multichannel device driver is not available from the sound card's manufacturer, you probably won't get a true surround-sound preview.) For sound cards that offer an interleaved multichannel driver, you should select this from the list. These driver types will accept the 6-audio input from Adobe Audition and automatically route it to the standard Microsoft 5.1 channel configuration.*

3 From the Preview Format Selector menu, select the bit rate of the preview playback material that is sent to your sound card. If your session includes higher bit rate files, and if your sound card supports it, you can select a higher rate to the preview your session more accurately.

4 Set the Preview Buffer Size for the buffers used for Play Track and Play All. Larger buffer sizes enable a more stable preview playback, but increase the latency (that is, makes it take longer to play the result of changes made while previewing). If dropouts occur when you preview the audio, try increasing the buffer size.

Exporting surround-sound files

Adobe Audition includes the ability to encode directly to an interleaved 6-channel Windows Media 9 Pro (WMA) file or to export into two WAV formats. The Format Options field in the Multichannel Export Options dialog box indicates the currently selected format. The format is retained from your last used export option.

Note: *To export and encode your project to a 6-channel WMA file, you must have Windows Media 9 runtime installed (it is installed by default as part of the Adobe Audition installation). If you have an earlier version installed, the Encode to WMA9 option will not be available. The latest Windows Media Updates are available on the Microsoft Web site.*

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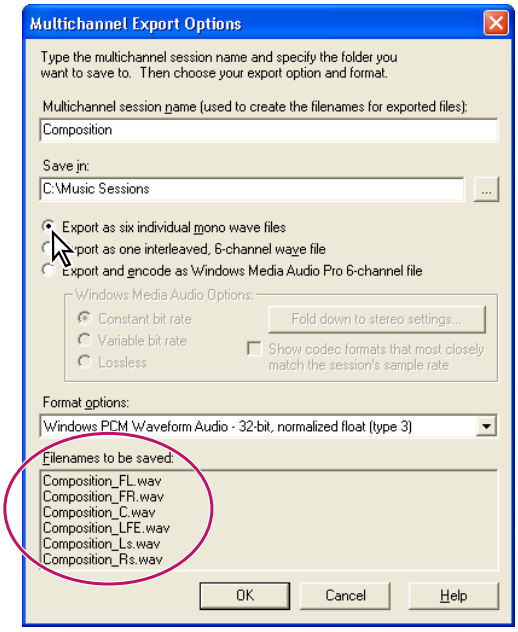
ug.book Page 224 Tuesday, March 16, 2004 1:29 PM


Multichannel Export Options
Type the multichannel session name and specify the folder you want to save to. Then choose your export option and format.
Multichannel session name (used to create the filenames for exported files):
Composition
Save in:
C:\Music Sessions
Export as six individual mono wave files
Export as one interleaved, 6-channel wave file
Export and encode as Windows Media Audio Pro 6-channel file
Windows Media Audio Options:
Constant bit rate
Variable bit rate
Lossless
Fold down to stereo settings...
Show codec formats that most closely match the session's sample rate
Format options:
Windows PCM Waveform Audio - 32-bit, normalized float (type 3)
Filenames to be saved:
Composition_FL.wav
Composition_FR.wav
Composition_C.wav
Composition_LFE.wav
Composition_Ls.wav
Composition_Rs.wav
OK
Cancel
Help

The Multichannel Export Options dialog box

Exporting a multichannel session

Once you've completed mixing your multichannel project, you can export it to your desired file format.

To export your multichannel session:

1 Click Export at the bottom right of the Multichannel Encoder dialog box.

2 In the Multichannel Session Name text box, enter a name for your exported files. You can see the names for all the files that will be saved at the bottom of the dialog box in the Filenames To Be Saved area.

3 In the Save In text box, enter or navigate to the directory to which you want to save the files.

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ug.book Page 225 Tuesday, March 16, 2004 1:29 PM

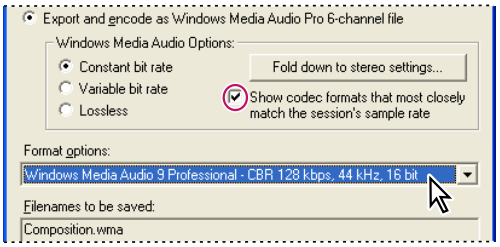
4 Select one of the following:

Export As Six Individual Mono Wave Files Creates standard Windows PCM .wav mono files that typically can be used by any Windows audio application. (For more information, see "Windows PCM (.wav)" on page 239.)

Export As One Interleaved, 6-Channel Wave File Exports as Windows PCM .wav format, which allows a single file to contain multiple channels of audio. However, not all Windows audio applications can open or play WAV files that are not mono or stereo. (For more information, see "Windows PCM (.wav)" on page 239.)

Note: *Interleaved files reflect the channel order used by Dolby Digital encoders. If you plan to use an encoding process with a different channel order, export the session as six individual files, instead.*

Export And Encode As Windows Media Audio Pro 6-Channel File Creates multichannel WMA files that can be played by anyone who has Windows Media Player 9 or later, a multichannel output sound card, and a 5.1 speaker setup. (Media Player 9 requires Windows XP.)


Export and encode as Windows Media Audio Pro 6-channel file
Windows Media Audio Options:
Constant bit rate
Variable bit rate
Lossless
Fold down to stereo settings...
Show codec formats that most closely match the session's sample rate
Format options:
Windows Media Audio 9 Professional - CBR 128 kbps, 44 kHz, 16 bit
Filenames to be saved:
Composition.wma

Export options

5 If you select Export And Encode As Windows Media Audio Pro 6-Channel File, specify the following Windows Media Audio options:

- Constant Bit Rate (CBR) varies the quality level as needed to ensure that the bit rate stays the same. This method makes a consistently sized file, although the quality may not be as high as with Variable Bit Rate encoding.
- Variable Bit Rate (VBR) maintains the audio quality by varying the bit rate depending on the complexity of the audio passage being encoded. This method can maintain higher quality audio in the file, although the file size is not as predictable as with Constant Bit Rate encoding
- Lossless compresses to a smaller file size than WAV, but results in no fidelity loss.

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ug.book Page 226 Tuesday, March 16, 2004 1:29 PM

- Fold Down To Stereo Settings folds down the 6-channel playback to a stereo playback on a non-Windows XP system or a system without a 5.1 playback setup. Specify the attenuation parameters to control how the levels of the Center, Surround, and LFE channels get mixed down with the front stereo channels and played back on a stereo output system. The defaults are usually good settings for most files, but you can enter any value in any of these three fields between 0 and –144 dB, as desired.
- Show Codec Formats That Most Closely Match The Session's Sample Rate limits the list of selectable WMA kbps options to those that are the same sample and bit rate as the multitrack session's files.

Exporting to a mastering or duplication service

If your project is to be sent out to a mastering, duplication, or other outside service with the intention of being encoded into other specific surround or media formats, check with the recipient as to the format specifics.

Channel ordering differs between surround formats, as do the crossover frequency points. For example, Digital Theater System (DTS) typically employs a crossover of 80 Hz, meaning that all frequency content of your channels lower than 80 Hz can be routed to a subwoofer, and all frequency content greater than 80 can be sent to the main channels. This differs from the Dolby Digital system that utilizes a crossover point of 120 Hz. Some systems also employ a boost of 10 dB for the LFE channel, automatically assuming your LFE content will be approximately this much lower in power than the main channels. Therefore, these components should be accounted for in your mix before you deliver your master files to the recipient. It is best to inquire with the recipient about all such requirements to ensure that the audience will hear your project the same way you are hearing it on your monitoring system.

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ug.book Page 227 Tuesday, March 16, 2004 1:29 PM

Chapter 11: Saving, Exporting, and Closing Files

Adobe Audition lets you save audio files and export sessions to a variety of common audio file formats.

Saving audio files

When working with audio files in Edit View, you can save your audio in a variety of common file formats. The format you choose depends on how you plan to use the file. For more information on supported file formats, see "Choosing an audio file format" on page 231.

When choosing a file format, keep in mind that different formats allow different information to be stored with the file. As a result, saving a file in a format different from its original format might cause some information to be discarded.

To save an audio file in Edit View:

1 Do one of the following:

- Choose File > Save to save changes you made to the current file. Alternatively, click the Save button in the File toolbar.
- Choose File > Save As to save changes to a different file. Alternatively, click the Save As button in the File toolbar.
- Choose File > Save Copy As to save an identical copy of the file while leaving the original file active.
- Choose File > Save Selection to save the currently selected audio to a new file. Alternatively, click the Save Selection button in the File toolbar. This command is useful for saving small segments of a larger file. For example, you can use it to break up a long recording into smaller, more manageable tracks.
- Choose File > Save All to save all open files.

2 Choose a location for the file, type a filename, and choose a file format.

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ug.book Page 228 Tuesday, March 16, 2004 1:29 PM

3 Depending on the format you choose, additional options might be available. To view format-specific options, click Options. For more information on format-specific options, see "Choosing an audio file format" on page 231.

4 Select Save Extra Non-Audio Information to save header fields containing file information and cue marks in the file. In addition, if you save a .wav file, this option stores the pathname to the original session file, effectively linking related session and mixdown files for Adobe Premiere and Adobe After Effects users. For more information, see "Working with Adobe Premiere Pro and After Effects" on page 207.

If you plan to burn the file to CD by using another program, you should deselect this option. Some CD recording applications interpret non-audio information incorrectly and place an unpleasant burst of noise at the beginning of each track.

5 Click Save.

Saving and exporting sessions

When you edit a session in Multitrack View, it's a good practice to save the session file frequently. After you create a mix, you can export the session to a variety of audio and video formats.

Saving sessions

The most important thing to remember about session (.ses) files in Adobe Audition is that they contain no audio data themselves. Instead, a session file is a small file that points to other audio files on your hard drive. The session file keeps track of where the audio files are stored on the hard drive, each file's location and duration within the session, what envelopes and effects are applied to the tracks, and so on.

A session file is useless without the audio files that it points to, so it's important to keep your files organized. The best way to stay organized is to keep all session-related files in the same folder. Adobe Audition makes organizing files easy by providing an option to save a copy of every file used in a session into the same folder as the session file. This option ensures that all files for a session are in one place.

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ug.book Page 229 Tuesday, March 16, 2004 1:29 PM

To save a session:

1 Do one of the following:

- Choose File > Save Session to save changes to the current session file. Alternatively, click the Save button in the Multitrack File toolbar.
- Choose File > Save Session As to save changes to a different session file. Alternatively, click the Save As button in the Multitrack File toolbar.
- Choose File > Save All to save all open sessions.

2 Choose a location for the file, and type a filename.

3 Select Save Copies Of All Associated Files to save a copy of every file used in a session into the same folder as the session file. It is highly recommended that you select this option.

If you want to save the associated files in a different format, click Options, select Save All Copies In This Format, and select a format from the list. To view options for the selected format, click Format Properties. For more information on format-specific options, see "Choosing an audio file format" on page 231.

4 Click Save.

To convert the sample rate of a session:

1 Choose File > Save Session As, choose a location for the file, and type a filename.

2 Select Save Copies Of All Associated Files, and click Options.

3 Select Convert Sample Rate, and select a sample rate.

4 To set dithering and other conversion options, click Conversion Properties. For more information on conversion options, see "Converting the sample type" on page 110.

5 Click Save.

Exporting mixes to audio

After you finish mixing a session, you can export it in a variety of common audio file formats. When you use the File > Export > Audio command, everything in the session is exported to an audio file. If you want to export only specific waves, use the Edit > Mix Down To File command instead. (See "Mixing down ReWire tracks and specific audio clips" on page 196.)

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ug.book Page 230 Tuesday, March 16, 2004 1:29 PM

To export a mix to an audio file:

1 Do one of the following:

- To export part of a session, select the desired area in the track display.
- To export an entire session, deselect everything in the track display. (If necessary, click the track display to reveal the current-time indicator.)

2 Choose File > Export > Audio.

3 Choose a location for the file, type a filename, and choose a file format.

4 Depending on the format you choose, additional options may be available. To view format-specific options, click Options. For more information on format-specific options, see "Choosing an audio file format" on page 231.

5 Select Save Extra Non-Audio Information to save header fields containing file information and cue marks in the file. In addition, if you save a .wav file, this option stores the pathname to the original session file, effectively linking related session and mixdown files for Adobe Premiere and Adobe After Effects users. For more information, see "Working with Adobe Premiere Pro and After Effects" on page 207.

If you plan to burn the file to CD by using another program, you should deselect this option. Some CD recording applications interpret non-audio information incorrectly and place an unpleasant burst of noise at the beginning of each track.

6 Click Save.

Exporting mixes to video

If a session includes an .avi video file, you can mix down the session and make it an audio track for the video.

Note: *While Adobe Audition can open other types of video files to get at their audio tracks, the ability to save back the audio track works only with .avi files.*

To export a mix to a video file:

1 Choose File > Export > Video.

2 Choose a location for the file, and type a filename.

3 To assign a codec for compressing the audio in the file, click Options. Choose a codec from the pop-up menu, and click OK.

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ug.book Page 231 Tuesday, March 16, 2004 1:29 PM

4 Select Save Extra Non-Audio Information to save header fields containing file information and cue marks in the file.

5 Click Save.

Closing files

Adobe Audition provides several commands for closing files.

To close the current audio file in Edit View:

Choose File > Close.

To close a session file in Multitrack View:

Do one of the following:

- Choose File > Close Session to close the current session file but leave related media files open.
- Choose File Close Session And Its Media to close the current session file and all related media files.

To close all files not related to the current session:

Choose File > Close Only Non-Session Media Files.

To close all open files:

Choose File > Close All.

Choosing an audio file format

Adobe Audition lets you open and save files in the formats described in this section. In most cases, you should save uncompressed audio in Windows PCM format, and you should save compressed audio in either mp3PRO® format or Windows Media Audio format. You'll need to use other formats only in special situations.

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ug.book Page 232 Tuesday, March 16, 2004 1:29 PM

Some formats provide options for saving audio data. Click Options in the Save As dialog box to access these options.

Note: *If you want to save files in a format that's not listed here, you may be able to do so by using an ACM Waveform codec. For more information, see "ACM Waveform (.wav)" on page 233.*

64-bit doubles (RAW) (.dbl)

This format uses 8-byte doubles in binary form—8 bytes per sample mono, or 16 bytes per sample stereo interleaved. The 64-bit doubles format has no header—it's purely audio data, just like the raw PCM format.

8-bit signed (.sam)

This format is popular for building MOD files, since audio in MOD files is 8-bit signed. Many MOD editors allow samples to be inserted from or exported to files in this format. Files with the .sam extension contain 8-bit signed raw data, and by default, they have no headers. The sample rate starts off as 22050 Hz, but you can change the sample rate after you open the file by choosing Edit > Adjust Sample Rate.

A/mu-Law Wave (.wav)

The A-Law and mu-Law WAV formats (CCITT standard G.711) are common in telephony applications. These encoding formats compress the original 16-bit audio to 8-bit audio (for a 2:1 compression ratio) with a dynamic range of about 13-bits (78 dB). While A-Law and mu-Law encoded waveforms have a higher signal-to-noise ratio than 8-bit PCM, they also have a bit more distortion than the original 16-bit audio. Still, the quality is higher than you would get with some 4-bit ADPCM formats.

Note: *Files saved in this format expand automatically to 16-bits when opened, so you shouldn't save 8-bit files in this format.*

Options Choose from the following:

- A-Law 8-bit is a slight variation of the standard mu-Law format and is found in European systems.
- mu-Law 8-bit is the international standard telecommunications encoding format and is the default option.

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ug.book Page 233 Tuesday, March 16, 2004 1:29 PM

ACM Waveform (.wav)

Microsoft ACM (Audio Compression Manager) is part of all 32-bit versions of Windows. Adobe Audition supports the ACM driver, which enables you to open and save files in a variety of formats other than those directly supported by Adobe Audition.

Some of these formats come as a standard part of Windows, while others are provided by third-parties. You may acquire ACM formats when you install other software.

To save a file in an alternate format by using the ACM driver, choose File > Save As, choose ACM Waveform as the file format, and click Options. You can select from among various quality levels, and each level will give you different options for formats and attributes.

Note: *The ACM driver you want to use might require that the file be in a specific format before saving it. For example, if you want to save a file in the DSP Group TrueSpeech format, you should first use the Edit > Convert Sample Type command to convert the file to 8 KHz, mono, 16-bit, because that is the only format that the TrueSpeech ACM driver supports. For more information on any particular ACM driver, contact the creator of the format (such as DSP Group for TrueSpeech, or CCITT for the various CCITT formats) or the manufacturer of the hardware that uses the format in question.*

Amiga IFF-8SVX (.iff, .svx)

The Amiga IFF-8SVX format is an 8-bit mono format from the Commodore Amiga computer.

Options Choose from the following:

- Data Formatted As saves the audio file in uncompressed 8-bit Signed format (the default setting) or in the compressed 4-bit Fibonacci Delta Encoded format.
- Dithering From 16-bit specifies a type of dithering for 16-bit files: Triangular Dither, Shaped Gaussian Dither, Noise Shaping A, or Noise Shaping B. No Dithering is the default. For more information on types of dithering, see "Changing the bit depth" on page 113.

Apple AIFF (.aif, .snd)

AIFF is Apple's standard wave file format. AIFF supports mono or stereo files, 16-bit or 8-bit resolution, and a wide range of sample rates. Adobe Audition supports only the PCM-encoded portion of the data, even though this format (like Windows WAV) can contain any one of various data formats.

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ug.book Page 234 Tuesday, March 16, 2004 1:29 PM

AIFF is a good choice for Windows/Mac OS cross-platform compatibility. Before you open AIFF files in Adobe Audition, add the .aif or .snd extension to the file and open it by using the Apple AIFF file filter. When you transfer an AIFF file to a Macintosh, you can add the four character code "AIFF" in the file's resource fork to have it recognized. (The Macintosh identifies a file through its "resource," which is removed when a file is opened on a Windows computer. However, many Mac OS applications that support AIFF can recognize the PCM data without this identifier.)

ASCII Text Data (.txt)

Audio data can be read to or written from files in a standard text format, with each sample separated by a carriage return, and channels separated by a tab character. An optional header can be placed before the data. If no header text exists, then the data is assumed to be 16-bit signed decimal integers. The header is formatted as a KEYWORD: value with the keywords being SAMPLES, BITSPERSAMPLE, CHANNELS, SAMPLERATE, and NORMALIZED. The values for NORMALIZED are either TRUE or FALSE. For example,

```
SAMPLES: 1582
BITSPERSAMPLE: 16
CHANNELS: 2
SAMPLERATE: 22050
NORMALIZED: FALSE
164 <tab> -1372
492 <tab> -876
```

Options Choose any of the following:

- Include Format Header places a header before the data.
- Normalized Data normalizes the data between –1.0 and 1.0.

Audition Loop (.cel)

This format produces compressed Adobe Audition loop files, which are essentially .mp3 files with a .cel extension. Each .cel file has a header that contains loop information, such as the number of beats, tempo, key, and stretch method.

You can also save loops in uncompressed formats, such as Windows PCM.

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ug.book Page 235 Tuesday, March 16, 2004 1:29 PM

The .cel format avoids a potential problem with .mp3 files. During encoding, a very small amount of silence is added to the beginning, end, or both of an .mp3 file. The silence is very short—often only a few samples long. When you work with a loop, though, it's enough to throw off the entire loop.

As it saves a .cel file, Adobe Audition calculates how much silence will be added to the .mp3 file and writes this information into the .cel header. Then, when Adobe Audition opens a .cel file, it reads this information and automatically removes the silence from the file so that it loops smoothly.

The options for Audition Loop format are identical to those for mp3PRO®. For more information, see "mp3PRO® (.mp3)" on page 237.

Creative Sound Blaster (.voc)

This format is for Sound Blaster and Sound Blaster Pro voice files. Adobe Audition supports both the older and newer formats. The older format supports only 8-bit audio, mono to 44.1 kHz and stereo to 22 kHz. The newer format supports both 8- and 16-bit audio.

Files in this format can contain information for looping and silence. If a file contains loops and silence blocks, they expand when you open the file.

Options Choose one of the following:

- Old Style saves audio as an 8-bit .voc file that can be played on any Sound Blaster card.
- New Style saves audio to the newer format that supports both 8- and 16-bit audio.

Dialogic ADPCM (.vox)

The Dialogic ADPCM format is used in telephony applications, and it's optimized for low sample rate voice. It supports only mono 16-bit audio, and like other ADPCM formats, it compresses the audio data to 4 bits/sample (4:1). This format has no header, so Adobe Audition assumes any .vox file is in Dialogic ADPCM format.

Note: *Take note of the sample rate of the audio before saving it, as you need to enter it upon reopening the file.*

DiamondWare Digitized (.dwd)

This format is used by DiamondWare Sound Toolkit, a programmer's library that lets you quickly and easily add high-quality interactive audio to games and multimedia applications. It supports both mono and stereo files at a variety of resolutions and sample rates.

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ug.book Page 236 Tuesday, March 16, 2004 1:29 PM

DVI/IMA ADPCM (.wav)

The International Multimedia Association (IMA) flavor of ADPCM compresses 16-bit data to 4 bits/sample (4:1) by using a different (faster) method than Microsoft ADPCM. It has different distortion characteristics, which can produce either better or worse results depending on the sample being compressed. As with Microsoft ADPCM, use this format with 16-bit rather than 8-bit files. This compression scheme can be a good alternative to MPEG; it provides reasonably fast decoding of 4:1 compression, and it degrades sample quality only slightly.

Options Choose from the following:

- 2 bits/sample, 8:1 produces files with the highest compression ratio (8:1) but with the lowest number of bits. Select this option if smaller file size is more important than audio quality. Keep in mind that this compression rate is less compatible than the standard 4-bit and is supported on fewer systems.
- 3 bits/sample, 5.3:1 produces higher quality than the 2 bits option, but the quality isn't quite as good as with the 4 bits and 5 bits options. Some systems might have problems playing back files with this compression rate, especially stereo files.
- 4 bits/sample, 4:1 produces 4-bit files at a compression ratio of 4:1. This option is the default.
- 5 bits/sample, 3.2:1 produces files with the highest quality, since more bits and a lower compression ratio are used. However, this compression rate is less compatible than the standard 4-bit.

Microsoft ADPCM (.wav)

The Microsoft ADPCM format provides 4:1 compression. Files saved in this format expand automatically to 16-bits when opened, regardless of their original resolution. For this reason, use this format with 16-bit rather than 8-bit files.

Options Choose from the following:

- Single Pass (Lower Quality) compresses files in a single pass. Use this option if you're in a hurry. However, the quality is lower than if you use the Multiple Pass option. The time taken to read an ADPCM-compressed file is the same no matter which option you use.
- Multiple Pass (Higher Quality) compresses files in multiple passes, providing better quality. This setting is the default.

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ug.book Page 237 Tuesday, March 16, 2004 1:29 PM

- Block Size offers three size options, each with a different compression ratio and quality level: Large (Default Quality), with a compression ratio of 3.98:1; Medium (Good Quality), with a compression ratio of 3.81:1; and Small (High Quality), with a compression ratio of 3.25:1.

mp3PRO® (.mp3)

The mp3PRO filter enables Adobe Audition to directly encode and decode .mp3 files. When you save a file to mp3 format, the audio is encoded and compressed according to the options you select. When you open an .mp3 file, the audio converts into the uncompressed internal format of Adobe Audition. As a result, you can save an .mp3 file in any format.

Avoid compressing the same audio to mp3 more than once. Opening and resaving an .mp3 file causes it to be recompressed, so any artifacts from the compressing process become more pronounced.

MP3/mp3PRO® Encoder Options dialog box contains two sets of options: basic options for choosing an encoding method and more advanced options. To view the advanced options, click Advanced. To view only the basic options, click Simple.

Basic options Choose from the following:

- CBR (constant bit rate) encodes the same bit rate throughout the entire file. This method is the most common and the most predictable for bandwidth and file size.
- VBR (variable bit rate) encodes higher bit rates for more complex material and lower bit rates for simpler material. While it depends on the source material, VBR-encoded .mp3 files generally tend to be smaller than CBR-encoded .mp3 files. Use the menu below the VBR option to choose a quality level from 10 (lowest quality but smaller file) to 100 (highest quality but larger file). Some mp3 players don't support VBR-encoded files. For maximum compatibility, select CBR.
- MP3 encodes the file to mp3, but without the PRO data.

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ug.book Page 238 Tuesday, March 16, 2004 1:29 PM

- mp3PRO® encodes the file mp3PRO. The PRO data helps re-create high frequencies in the compressed file, especially at low bit rates. An mp3PRO file can still be played back by an mp3 player that doesn't support the PRO data, but the quality may be lower than for a standard mp3 file of that bit rate. For example, a 64 Kbps mp3PRO file sounds more like a 112 Kbps or 128 Kbps mp3 file if the player supports mp3PRO, but it sounds like a 64 Kbps mp3 file (or worse) if the player doesn't support mp3PRO.

For information on advanced mp3PRO options, see "mp3PRO® (.mp3)" in Help.

NeXT/Sun (.au, .snd)

The NeXT/Sun format is standard on NeXT and Sun computers, and it has many data types. Adobe Audition supports the CCITT A-Law, mu-Law, G.721 ADPCM, and linear PCM data variants. Like Windows PCM and AIFF, this format can support mono or stereo, 16- or 8-bit, and a wide range of sample rates when saved as linear PCM.

The NeXT/Sun format is most commonly used for compressing 16-bit data to 8-bit mu-law data. AU is used quite extensively on the Web and in Java applications and applets.

Options Choose from the following:

- mu-Law 8-bit uses the mu-law 8-bit format to compress the file.
- A-Law 8-bit uses the A-law 8-bit format to compress the file.
- G.721 ADPCM 4-bit applies the standard CCITT G.721 compression to the file (ADPCM at 32Kbps).
- Linear PCM saves the file as uncompressed, linear PCM (Pulse Code Modulation).

SampleVision (.smp)

The SampleVision format is native to Turtle Beach's SampleVision program. This format supports only mono, 16-bit audio. If a file is in a different format, Adobe Audition prompts you to convert it before saving it.

This format also supports loop points, which you can edit in the Cue List window. The Label of the cue must be in the format `Loop n, m` where "n" is the loop number from 1 to 8, and "m" is the mode (0 = no looping, 1 = forward loop, 2 = forward/back loop). In the Play List window, you can enter the number of times to loop the cue range.

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ug.book Page 239 Tuesday, March 16, 2004 1:29 PM

Windows Media Audio (.wma)

The WMA format utilizes a perceptual compression scheme and lets you select from three different encoding options:

- Constant Bit Rate Encoding varies the quality level as needed to ensure that the bit rate stays the same. This method makes a consistently sized file, although the quality may not be as high as with Variable Bit Rate encoding.
- Variable Bit Rate Encoding maintains the audio quality by varying the bit rate depending on the complexity of the audio passage being encoded. This method can maintain higher quality audio in the file, although the file size is not as predictable as with Constant Bit Rate encoding.
- Mathematically Lossless Encoding compresses to a smaller file size than WAV, but results in no fidelity loss.

After you select an encoding option, you can set the desired quality. Just as with stereo WMA files, the higher quality setting you select, the larger the file size, and vice versa.

Windows PCM (.wav)

The Microsoft Windows PCM format supports both mono and stereo files at a variety of resolutions and sample rates. It follows the RIFF (Resource Information File Format) specification and allows for extra user-information to be embedded and saved with the file. The WAV format reproduces digital audio by using PCM (Pulse Code Modulation)—PCM doesn't require compression and is considered a lossless format.

Options The following options are available for 32-bit files; no options are available for 8- or 16-bit files:

- 32-bit Normalized Float (type 3) – Default is the internal format for Adobe Audition and the standard floating point format for type 3 .wav files. Values are normalized to the range of +/–1.0, and although values beyond this range are saved, clipping may occur in some programs that read them back in. (Adobe Audition won't clip audio but will instead read the same value back if it's beyond this range.)

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ug.book Page 240 Tuesday, March 16, 2004 1:29 PM

- 24-bit Packed Int (type 1, 24-bit) saves straight 24-bit integers so any data beyond the bounds is clipped. The .wav BitsPerSample is set to 24 and BlockAlign is set to 3 bytes per channel.
- 24-bit Packed Int (type 1, 20-bit) saves straight 24-bit integers so any data beyond the bounds is clipped. The .wav BitsPerSample is set to 20 and BlockAlign is set to 3 bytes per channel. The extra 4 bits are actually the remaining valid bits when saving, and they are used when reading (thus still giving 24-bit accuracy if those bits were actually present when writing). Applications either fill those last 4 bits with zeros or with actual data; analog/digital converters that generate 20 bits of valid data automatically set the remaining 4 bits to zero. Any type 1 format with BlockAlign set to 3 bytes per channel is assumed to be packed integers, and a BitsPerSample value between 17 and 24 will read in all 24 bits and assume the remaining bits are either accurate or set to zero.
- 32-bit 24.0 Float (type 1, 24-bit) – Non-Standard saves full 32-bit floats (ranging from +/–8million), but the .wav BitsPerSample is set to 24 while BlockAlign is still set to 4 bytes per channel.
- 16.8 float – Obsolete/Compatibility is the internal format used by Adobe Audition 1.0. Floating point values range from +/–32768.0, but larger and smaller values are valid and aren't clipped since the floating point exponent is saved as well. The .wav BitsPerSample is set to 32 and BlockAlign is set to 4 bytes per channel.
- Enable Dithering dithers 32-bit files when they are saved to a PCM format (20-bit, 24-bit, or 32-bit). This option is available only for a 32-bit file that you select to save to a nonfloating-point type format. It applies a Triangular dither with a 1.0 depth 1.0 and no noise shaping. If you wish to apply a noise-shaped dither, use the Edit > Convert Sample Type command to dither the audio first, and then save the file without dithering enabled in the file format options.

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ug.book Page 241 Tuesday, March 16, 2004 1:29 PM

PCM Raw Data (.pcm) (.raw)

This format is simply the PCM dump of all data for the wave. No header information is contained in the file. For this reason, you must select the sample rate, resolution, and number of channels upon opening the file.

By opening audio data as PCM, you can interpret almost any audio file format—but make sure that you have some idea about the sample rate, number of channels, and so on. You can also interpret the data as A-law or mu-law compressed. When you guess at these parameters upon opening a file, it may sound incorrect (depending on which parameters are wrong). Once the file is opened and sounds fine, you may hear clicks at the start or end of the waveform, or sometimes throughout. These clicks are various header information being interpreted as waveform material. Just cut these out, and you've read in a wave in an unknown format.

Options Choose from the following:

- Data Formatted As specifies the format of the saved data.
- When Opening, Offset Input Data By specifies the number of bytes by which to offset the input data.
- Create .DAT Header File On Save writes a header to a separate .dat file to make reopening the file easier.

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ug.book Page 243 Tuesday, March 16, 2004 1:29 PM

Chapter 12: Scripting and Batch Processing

If you regularly perform certain audio tasks, you can automate them with scripts and batch processing to work more efficiently.

About scripting and batch processing

Adobe Audition scripts let you save a series of actions such as copying data or applying an effect, so you can perform those actions again with the click of a button. Scripts are simple text files that are similar to macros; Adobe Audition stores the exact actions of your mouse and any tweaking of parameters, so you can repeat them in the same sequence when you run the script.

For example, suppose you have a combination of effects with particular settings (an EQ setting, a Hall reverb, and so on) that you want to apply often and in combination to achieve a certain sound. You can record these steps, along with effects' specific settings, and then apply them at any time simply by calling the script.

Batch processing cue ranges

You can use the Batch feature in the Cue List to add silence between cues and save the audio between cues to new files. For more information about cues, the Cue List, and cue ranges, see "Working with cues" on page 96.

To batch process cues:

1 Choose Window > Cue List.

2 Select one or more cues in the Cue List dialog box. At least one of the cues you select must be a range.

3 Click Batch at the bottom of the dialog box.

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ug.book Page 244 Tuesday, March 16, 2004 1:29 PM

4 Set the following options as desired, and click OK:

Set Amount Of Silence Adds silence between cue points in the current waveform. Enter the number of seconds of silence you want in the Add Silence Before and Add Silence After text boxes.

Save To Files Splits the audio between cue points in the active waveform to new files.

Use Cue Label As Filename Uses the name of the cue as the prefix for the filename.

Filename Prefix Specifies the prefix for the filename (such as "phrase"). Adobe Audition automatically adds numbers after the prefix (phrase02, phrase03, and so on) in addition to the correct extension based on the output format you specify.

Seq. Start Specifies the number to begin with when appending numbers to the filename prefix.

Destination Folder Specifies the folder in which you want Adobe Audition to place new "split" files. Click Browse to open the Choose Destination Folder window and locate a different folder.

Output Format Sets the output format. Depending on the format, Options is available. Click Options to select options for that format.

Normalizing groups of files

When you normalize a waveform, the loudest part of the waveform is set to a specified amplitude, thereby raising or lowering all other parts of the same waveform by the same amount. Group Waveform Normalize lets you normalize the volume of multiple open waveforms by using a three-screen batch process. If the volume is raised as part of the normalization process, Adobe Audition can apply limiting to prevent clipping.

If you're getting ready to master an audio CD, using Group Waveform Normalize is a great way to make sure that all tracks on the CD have a consistent volume.

To normalize a group of files:

1 Choose Edit > Group Waveform Normalize.

2 Select the open waveforms you want to normalize. Click to select a single file, Shift-click to select contiguous files, Ctrl-click to select noncontiguous files, and drag to select a group of files.

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ug.book Page 245 Tuesday, March 16, 2004 1:29 PM

3 Click the Analyze Loudness tab, and then click Scan For Statistical Information to display amplitude statistics for each waveform. Double-click a file in this list to see more detailed statistics, including an RMS histogram and a clipping profile. (See "Understanding statistics on the Analyze Loudness tab" on page 245.)

4 Click the Normalize tab, and specify how you want to normalize the waveforms. (See "Setting options on the Normalize tab" on page 246.)

5 Click Run Normalize.

Understanding statistics on the Analyze Loudness tab

When you click Scan For Statistical Information, the Analyze Loudness tab displays the following information:

Eq-Loud Is the final loudness value with an equal-loudness equalization curve that takes into account frequencies to which the human ear is most sensitive. If you select the Use Equal Loudness Contour option in the Normalize tab, this value determines how much to amplify the audio to normalize it.

Loud Is the final loudness value without equal-loudness equalization. If you don't select the Use Equal Loudness Contour option in the Normalize tab, this value determines how much to amplify the audio to normalize it.

Max Is the maximum RMS (Root-Mean-Square) amplitude present. This value is based on a full-scale sine wave being 0 dB, and it conforms to the width specified in the Advanced section of the Normalize tab.

Avg Is the average RMS of the entire waveform. This value isn't used for normalization.

% Clip Is the percentage of the waveform that would be clipped as a result of normalization. Clipping won't occur if limiting (in which loud passages are decreased in volume) is used; instead, the louder portions of audio are limited to prevent clipping. In general, avoid values higher than 5% to prevent audible artifacts from occurring in the louder portions of audio.

Reset Clears all of the normalization statistics for the files in the list.

Double-click a file in this list to see more detailed statistics, including a complete RMS histogram, which shows the relative amounts of audio at each loudness level, and a clipping profile, which shows how much clipping will occur for each decibel of amplification.

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ug.book Page 246 Tuesday, March 16, 2004 1:29 PM

Setting options on the Normalize tab

Use the following options in the Normalize tab to specify how you want to normalize the waveforms:

Normalization Specifies whether to normalize to an average level or a specific level you enter in decibels.

Note: *The Normalization option doesn't use percentages, unlike the Normalize effect, because it is RMS-based rather than peak-based.*

Use Equal Loudness Contour Applies an equal loudness contour, where the middle frequencies are most important. Because the human ear is much more sensitive to frequencies between 2 kHz and 4 kHz, two different pieces of audio with the same RMS amplitude but with different frequencies will have different apparent volumes. Select this option to ensure that audio has the same perceived loudness, regardless of what frequencies are present.

Out of Band Peaks Determines how Adobe Audition handles out-of-band peaks. When you amplify audio, the audio samples may extend beyond the clipping point. If out-of-band peaks occur, you can choose to just let it clip the waveform (and cause distortion), or you can apply limiting to those areas so the audio doesn't clip the waveform (a common practice for TV commercials so they sound louder).

- No Limiting (Clip) prevents limiting, so clipped (distorted) audio might occur.
- Use Limiting applies the Hard Limiter, if needed, to keep out-of-band peaks from being clipped. This options provides two additional options: Look Ahead Time and Release Time.
- Lookahead Time specifies the number of milliseconds generally needed to attenuate audio before reaching the loudest peak.

Note: *If this value is too small, audible distortion might occur. Make sure that the value is at least 5 milliseconds.*

- Release Time specifies the number of milliseconds needed for the attenuation to rebound 12 dB (or roughly the time needed for audio to resume normal volume if an extremely loud peak is encountered).

Note: *A setting of 200 milliseconds works well to preserve low bass frequencies. If the setting is too high, audio may stay quiet and not resume normal levels for a while.*

Statistics RMS Width Specifies the length of the audio selection to use for calculating the RMS (Root-Mean-Square) minimum and maximum values.

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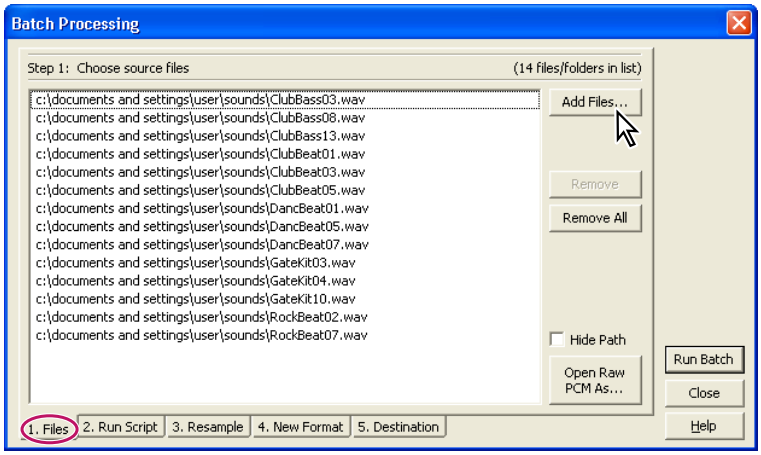
ug.book Page 247 Tuesday, March 16, 2004 1:29 PM

Batch processing files

The Batch Processing dialog box in Adobe Audition enables you to run a single script repeatedly over a group (batch) of source files.

Note: *For a script to run on a batch of files, you must record it in Script Works On Current Wave mode. That is, before you record the script, a waveform must be open with no selection made.*

In addition, the Batch Processing dialog box enables you to change multiple waveforms from one audio format to another (such as from WAV to MP3). For more information, see "Converting the sample type" on page 110.



The Files tab of the Batch Processing dialog box

To batch process files:

1 In Edit View, choose File > Batch Processing. The Batch Processing dialog box appears with the Files tab displayed.

2 Click Add Files to open the Please Choose The Source Files dialog box, and select one or more files:

- Hold down Ctrl or Shift to select noncontiguous or contiguous files, respectively.
- Click Remove to delete highlighted files from the list.
- Click Remove All to delete all files from the list.

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ug.book Page 248 Tuesday, March 16, 2004 1:29 PM

- Click Hide Path to display the name of the file without its full path.
- Click Open Raw PCM As to select the desired Sample Rate, Channels, Resolution, and other properties. Use this option only when converting Raw PCM files.

3 Click the Run Script tab at the bottom of the Batch Processing dialog box.

4 Select Run A Script. Then, click Browse to locate and select a script collection (*.scp) file, and click Open.

5 Choose a script from the Script menu. The only scripts that you can use for batch processing (and the only ones that appear in the list) are those that were recorded in Script Works On Current Wave mode. (See "Working with scripts" on page 249.)

6 Click the Resample tab.

7 Select Conversion Settings to change each waveform's sample properties to a common set of values. Then, click Change Destination Format to specify the values. (See "Converting the sample type" on page 110.) If you don't select Conversion Settings, the sample properties for the destination files are the same as those for the source files.

8 Click the New Format tab.

9 From the Output menu, choose a format for the destination files.

10 Click Format Properties to display options for the destination format.

Note: *Sample Format Types lists the sample properties of the waveforms that are to be converted. If more than one entry is listed, you might have to select different properties for each, depending on the destination format. For instance, a 22 kHz mono waveform might need different encoding options than a 44 kHz stereo file.*

11 Click the Destination tab.

12 Select a destination folder, specify how files are renamed by setting the following options, and then click Run Batch:

Same As File's Source Folder Saves modified files in the same folder as the file's source file.

Other Folder Specifies the folder in which to save modified files. Click Browse to locate a folder.

Overwrite Existing Files Saves existing files with a new name.

Delete Source File If Converted OK Deletes source files after they are converted successfully.

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ug.book Page 249 Tuesday, March 16, 2004 1:29 PM

Remove From Source List If Converted OK Removes filenames from the source list after the files are converted successfully.

Output Filename Template Specifies how files are renamed. By default, the first part of the filename remains the same, and the extension changes to match the chosen output format. Alternatively, you can type a different extension, and you can set up conditions for how files are renamed by using question marks and asterisks:

- A question mark ("?") signifies that a character doesn't change.
- An asterisk ("*") denotes the original filename or extension.

Here are some examples of how filenames can be renamed:

Original Name	Output Filename Template Name	Resulting Filename
zippy.aif	*.wav	zippy.wav
toads.pcm	q*.voc	qtoads.voc
funny.mp3	b???????.*	bunny.mp3
biglong.au	????.au	bigl.au
bart.wav	*x.wav	bartx.wav

Working with scripts

Adobe Audition lets you create three types of scripts, depending on the software's state when you record the script:

- Scripts that start from scratch. These scripts start with no waveform opened, and their first command is File > New.
- Scripts that work on the currently open waveform. These scripts operate on an entire waveform. They require a waveform to be open, but with no selection made. Actions begin at the current-time indicator position in the waveform, and they affect any data present at that point.
- Scripts that work on a selection. These scripts require a selection to be made first. Actions in the script apply only to the selection.

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ug.book Page 250 Tuesday, March 16, 2004 1:29 PM

A set of scripts can be grouped together in a *script collection*. For example, a script collection called "ambiance" might contain scripts for adding echo, reverb, and delay, and one called "batch utilities" might contain scripts for batch processing. (See "Batch processing files" on page 247.)

Creating scripts

Use the Scripts dialog box to create your scripts.

To create a script:

1 Set up Adobe Audition for the script you want to create. For example, open a waveform typical of the ones you'll apply the script to, or, if you want a script that starts from scratch, close all open waveforms.

2 In Edit View, choose Options > Scripts. The Script Collections area displays the name of the currently opened script collection. If the collection hasn't been named, the name New Collection appears.

3 Do one of the following:

- To open an existing script collection, click Open/New Collection, navigate to the collection (*.scp) file, and then double-click it.
- To create a new script collection, click Open/New Collection. Navigate to the folder in which you want to save the new collection (*.scp) file. Then, type a name for it in the File Name text box, and click Open.
- To rename a script collection, click Edit Script File. The collection (*.scp) file opens in Windows Notepad. Locate the "Collection:" entry on the first line, and type a new name. Then, save the file.

Note: *The name in the Script Collections area doesn't reflect the change until you reopen the script.*

4 Type a name for your script in the Title text box.

5 Click Record. The Scripts dialog box closes.

6 Perform the actions that you want to be part of the script.

Note: *Don't open or save a file as part of the actions for the script, since these actions are specific to a particular file. If you make a mistake, return to the Scripts dialog box, click Stop Current Script, click Clear, and start over.*

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ug.book Page 251 Tuesday, March 16, 2004 1:29 PM

7 After you record the script, choose Options > Scripts.

8 Click Stop Current Script.

9 Type a description for the script in the text area of the dialog box. The description appears when the script is selected.

Note: *You can add or edit a description later by clicking Edit Script File.*

10 Click Add to Collection. The script appears in the list at the left.

Running scripts

After you create a script, you can run it on a file, an entire waveform, or part of a waveform, depending on the script type.

To run a single script on a batch of files, use the Batch Processing command.

To run a script:

1 Set up Adobe Audition to match the starting point of the script. For example, if you want to run a script intended for a waveform, open a file and select a waveform. If you want to run a script that starts from scratch, close all open waveforms.

2 Switch to Edit View, and choose Options > Scripts. The Script Collections area displays the name of the currently opened script collection. If the collection hasn't been named, the name New Collection appears.

3 If the script collection you want isn't open, click Open/New Collection. Navigate to the collection (*.scp) file you want, and double-click it.

4 Select the script you want to run from the list.

5 Set the following options as desired, and then click Run Script:

Pause At Dialogs Stops the script at each dialog box used in the script, so you can modify the settings at those points. Clicking Cancel in any dialog box stops the script, and clicking OK continues it.

Alert When Complete Displays a notice when the script is finished.

Execute Relative To Cursor When running a Works On Current Wave type of script, performs all script operations relative to the original position of the cursor, as opposed to at the current position.

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ug.book Page 252 Tuesday, March 16, 2004 1:29 PM

For example, if a script was recorded with the current-time indicator at 0:10:00, selecting this option applies the script at the current cursor position, plus 10 seconds: If the current cursor position is at 0:05:00, the script would start at 0:15:00.

If you're likely to run a script at the current cursor position, record the script with the cursor at a 0:00:00 position, and select this option when you run it.

Script Type Indicates the type of script selected in the scripts list: Script Starts From Scratch works with all files closed; Script Works On Current Wave works on an entire waveform; and Script Works On Highlighted Section works on the selected part of a waveform.

Editing scripts

The Edit Script File option in the Scripts dialog box lets you modify existing scripts as text in a Windows Notepad file.

To edit a script:

1 In Edit View, choose Options > Scripts. The Script Collections area displays the name of the currently opened script collection. If the collection hasn't been named, the name New Collection appears.

2 If the script collection you want isn't open, click Open/New Collection. Navigate to the collection (*.scp) file you want, and double-click it.

3 Select the script you want to edit from the list.

4 Click Edit Script File. The collection file opens in Windows Notepad.

5 Scroll through the file to find the script you want.

6 Make the changes you want, and save the file.

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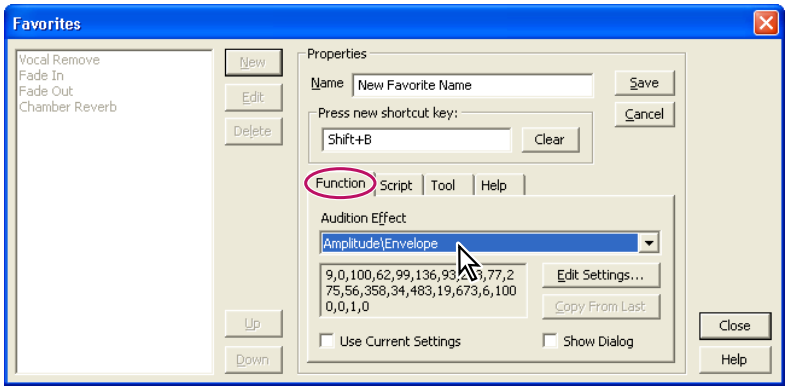
ug.book Page 253 Tuesday, March 16, 2004 1:29 PM

Using favorites (Edit View only)

The Favorites menu in Edit View lists custom commands you can create. The Favorites dialog box lets you create, edit, customize, and save these commands, which are based on your favorite Adobe Audition effects, scripts, and even third-party tools (the latter using command line executables). You can even organize favorites into hierarchical submenus for easy navigation.



The four tabs in the Favorites dialog box

To apply favorites:

In Edit View, choose Favorites, followed by the favorite you want to apply.

To create or edit favorites:

1 Choose Favorites > Edit Favorites.

2 Select from the following options, click Save, and then click Close:

New Enables the fields in the Properties area of the Favorites dialog box.

Edit Enables the fields in the Properties area for the selected favorite.

Delete Removes the selected favorite.

Up Moves the selected favorite up in the list. The Favorites menu reflects the order of the list.

Down Moves the selected favorite down in the list.

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ug.book Page 254 Tuesday, March 16, 2004 1:29 PM

Name Specifies the name of a favorite. Use this text box to help organize the Favorites menu by doing one or more of the following:

- Create hierarchical menus by using a backslash ("\"). For example, type **My Delays\Hall Reverb** in the Name text box to place the Hall Reverb favorite in the My Delays submenu.
- Add separator bars by typing a series of dashes ("------") into the Name text box. If you want more than one separator bar, type a different number of dashes, or add text so that the separator doesn't match one in the list. For example, type "------2" (the "2" after the dashes doesn't appear in the Favorites menu).
- Create a separator bar for a submenu by entering the submenu path first (such as "My Effects\------"). (The text that appears with a separator bar is for appearance only.)

***Note**: If you create text for a submenu title, make sure not to specify any command, script, or tool listed on the Function tab, Script tab, or Tool tab.*

Press New Shortcut Key Lets you type a key or combination of keys to use as the keyboard shortcut to a favorite. Adobe Audition accepts most single key shortcuts (the most notable exceptions are the Print Screen, Scroll Lock, Number Lock, Caps Lock, Tab, Function, and Enter keys), and it also accepts the Ctrl, Shift, and Alt keys (or any combination of the three) as the first in a combination of keys.

***Note:** If the keyboard shortcut you type is already used by Adobe Audition, a dialog box appears, giving you the option to overwrite the current shortcut.*

Clear Clears text from the Press New Shortcut Key text box.

Function tab Lets you specify the following options:

- Audition Effect lets you choose any command listed in the Effects and Generate menus. After you choose a command, the settings last used for it appear.
- Edit Settings displays the window that corresponds to the command you chose. You can then specify the settings to be used when you choose the favorite from the Favorites menu.
- Copy From Last applies the settings used the last time the particular command was completedsuccessfully.
- Use Current Settings applies the settings currently specified for the particular command. Deselect this option to edit the settings.

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ug.book Page 255 Tuesday, March 16, 2004 1:29 PM

- Show Dialog causes the dialog box for the particular command to display, with the settings you specified for the favorite.

Script tab Lets you specify the following options:

- Script Collection File displays the current script collection in use. The button to the right of the text box opens the Browse For Script dialog box that lets you navigate to and select a script collection (*.scp) file.
- Script lets you choose the script you want to run from the selected collection.
- Pause At Dialogs stops the script at each dialog box used in the script, so you can modify the settings at those points. Otherwise, the script runs nonstop to completion.

Tool tab Specifies the command line for the tool you want to run, including any command line switches the particular tool may need. The button to the right of the text box opens the Browse For Tool dialog box that lets you navigate to the desired tool.

Help tab Displays instructions for adding separators and submenus to the Favorites menu.

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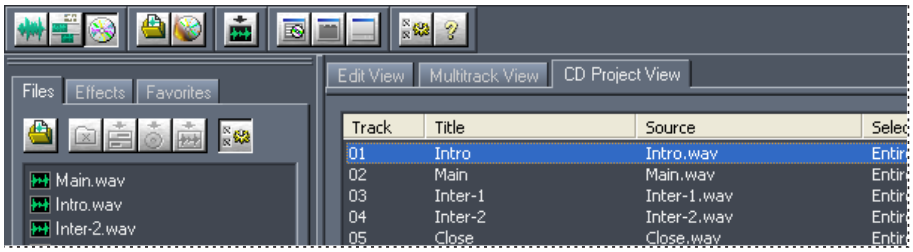
ug.book Page 257 Tuesday, March 16, 2004 1:29 PM

Chapter 13: Burning Audio CDs

Adobe Audition provides an integrated CD burning utility that makes it easy to create audio CDs from audio and session files.

Using CD Project View

CD Project View provides an easy-to-use interface for assembling CD tracks, setting track properties, and burning CDs. The display window in CD Project View contains the track list, which displays information about the audio tracks you assemble. CD Project View also shares many elements with Edit View and Multitrack View, such as dockable windows, menus, toolbars, and a status bar. (See "About the work area" on page 9.)



CD Project View

You use the tabs above the display window, commands in the View menu, or buttons in the toolbar to switch between views. (See "Switching between views" on page 11.)

Assembling tracks

You can assemble the tracks for a CD all at once, or you can insert individual tracks as you finish editing the audio. After you insert tracks, you can also change their order or remove them.

When you assemble audio for a CD, you'll probably want to fine tune the individual tracks so that they form a cohesive whole. This process—known as *mastering*—often involves cropping files, adjusting dynamics (compressing), and comparing the audio for continuity levels and EQ. (See "About the mastering process" on page 117 and "Normalizing groups of files" on page 244.)

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ug.book Page 258 Tuesday, March 16, 2004 1:29 PM

Inserting tracks

Adobe Audition provides a variety of ways to insert tracks into CD Project View. Keep in mind that you're not limited to inserting entire files—you can also insert audio ranges that are defined as track cues. For more information on creating track cues, see "Defining and selecting cues" on page 96 and "Choosing a cue type" on page 97.

To insert a track:

Do one of the following:

- Select one or more files or track cues in the Files tab of the Organizer window. Then, drag the items into the track list or click the Insert Into CD Project button . For more information on using the Files tab of the Organizer window, see "Organizing files" on page 24.
- In CD Project View, choose Insert > Audio or Insert > Audio From Video. Select a file, and click Open.
- In CD Project View, choose Insert > File/Cue List. Select the file or track cue you want to insert.
- Drag any supported audio file type from your desktop (Windows, My Computer, or Windows Explorer) directly into the track list in CD Project View. The file first opens in Adobe Audition, and then is inserted into the track list.
- In Edit View, open a file. To insert the entire file, make sure that no audio is selected; to insert part of a file, select the desired range. Then choose Edit > Insert In CD Project.
- In Multitrack View, open a session file, and choose Edit > Mix Down To CD Project. If the session includes track cues, each cue range is inserted into the track list as a separate track.

If you want to divide a single, long audio file (such as a recording of a concert that includes several songs) into multiple tracks on a CD, insert the file into a session, and add track cues at the desired locations. Then, choose Edit > Mix Down To CD Project. The cue ranges are inserted automatically as separate tracks.

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ug.book Page 259 Tuesday, March 16, 2004 1:29 PM

Selecting tracks

In CD Project View, you can select one or more tracks by clicking in the track list. You can also select all tracks by choosing the Select All Tracks command.

To select a track:

In CD Project View, click the track in the track list.

To select multiple tracks:

Do one of the following:

- To select adjacent (contiguous) tracks, click the first track in the desired range, and then Shift-click the last.
- To select nonadjacent (incontiguous) tracks, Ctrl-click them.

To select all tracks:

Choose Edit > Select All Tracks.

Rearranging tracks

In CD Project View, you can move tracks up and down to change their play order on a CD.

To rearrange tracks:

1. In CD Project View, select the track you want to move.
2. Click Move Up or Move Down.

Removing tracks

In CD Project View, you can remove a single track, multiple tracks, or all tracks.

To remove tracks:

In CD Project View, do one of the following:

- Select one or more tracks, and click Remove. Alternatively, choose Edit > Remove Selected Tracks.
- To remove all tracks, click Remove All. Alternatively, choose Edit > Remove All Tracks.

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ug.book Page 260 Tuesday, March 16, 2004 1:29 PM

To close the source files when removing tracks:

1 Select one or more tracks.

2 Choose Edit > Destroy Selected Tracks (Remove and Close).

Editing the source audio for tracks

The Edit Waveform command in CD Project View lets you edit the source audio for a track in Edit View.

To edit the source audio for a track:

1 In CD Project View, select the track you want to edit.

2 Choose Edit > Edit Waveform.

Setting track properties

Adobe Audition lets you specify a title and artist for each track. CD players that support CD Text display the text during playback.

You can also change the length of pauses between tracks, enable or disable copy protection and pre-emphasis features, and add an ISRC (International Standard Recording Code) number.

To set track properties:

1 In CD Project View, select the track for which you want to set properties, and click Track Properties. Alternatively, select the track, and choose View > Track Properties.

2 Enter a track title and artist for the track.

Important: *In order for Adobe Audition to write text to the CD, you must select Write CD-Text in the Write CD dialog box. (See "Writing a CD" on page 261.)*

3 If you want to set additional properties for the track, select Use Custom Track Properties. Set any of the following options, and click OK:

Pause Adds a pause of the specified length before the track. By default, Adobe Audition assigns a 2-second pause to the beginning of each track.

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ug.book Page 261 Tuesday, March 16, 2004 1:29 PM

Copy Protection Sets the copy protection flag (as defined by the Red Book specification) for the track. In order for copy protection to occur, the CD player must support the copy protection flag.

Pre-Emphasis Sets the pre-emphasis flag (as defined by the Red Book specification) for the track. *Pre-emphasis* is a basic noise reduction process that is implemented by a CD player. For pre-emphasis to occur, the CD player must support the pre-emphasis flag.

ISRC Specifies an ISRC (International Standard Recording Code). This code is used only on CDs that are destined for commercial distribution. ISRC codes have 12 characters and use the following format:

- ISO Country: 2 digit code (for example, *US* for USA).
- Registrant code: 3 digit alpha-numeric, unique reference.
- Year of reference: last 2 digits of the year (for example, *04* for 2004).
- Designation code: a 5 digit, unique number.

Same For All Tracks Applies the settings, with the exception of the ISRC code, to all tracks in the track list.

Writing a CD

Before you write a CD, you should verify that your CD burning device is set up correctly. Then, set CD options and write the CD.

Note: *Audio on CDs must be 44.1 kHz, 16 bit, stereo. If you insert a track with a different sample type, Adobe Audition automatically converts the audio for you.*

To set CD device properties:

1 In CD Project View, choose Options > Device Properties.

2 Select the device you want to set up.

3 Select a buffer size and write speed for the device.

4 If the device supports buffer underrun protection, select Buffer Underrun Protection to allow the drive to stop and resume burning as needed.

5 Click OK.

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ug.book Page 262 Tuesday, March 16, 2004 1:29 PM

To set CD options and write a CD:

1 Insert a blank, writable CD into the CD burning device.

2 In CD Project View, click Write CD or choose File > Write CD.

3 Choose the device you want to use to write the CD. (Click Device Properties to set device properties, as described in the previous procedure.)

4 Choose a setting from the Write Mode pop-up menu:

- Write CD writes the CD without testing for buffer underruns.
- Test Write Only tests if the CD can be written without the occurrence of buffer underruns. No audio is written to the CD.
- Test and Write CD tests for buffer underruns and then proceeds with the actual write process if the test is successful.

5 Select Eject Disc When Complete to eject the CD tray upon completion of the write process.

6 Select Write CD-Text if you want to write text, including the track title and artist for each track, to the CD. Type the desired information in the text boxes for Title, Artist, and UPC/EAN.

Note: *The UPC/EAN is a 13-digit code that is used to uniquely identify merchandise and communicate product information between a vendor and retailer.*

7 Click Write CD. The Track and Disk bars show you the progress of the write process.

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ug.book Page 263 Tuesday, March 16, 2004 1:29 PM

Appendix A: Keyboard Shortcuts

Keyboard shortcuts help you work more efficiently.

About keyboard shortcuts

The default keyboard shortcuts address most audio production needs, but you can also create custom shortcuts tailored to your working style. To customize shortcuts or trigger commands with a MIDI keyboard, use the Keyboard Shortcuts & MIDI Triggers command. (See "Using shortcuts" on page 12.)

Note: *Adobe Audition displays most default keyboard shortcuts in menu commands and tool tips. The user guide and Help list only shortcuts that Adobe Audition doesn't display.*

Keys for playing audio

Space	Toggle between Play and Stop
Ctrl+Space	Toggle between Record and Pause
Ctrl+Shift+Space	Toggle between Play All and Pause
Alt+O	Play postroll
Alt+R	Play preroll and postroll (skip selection)
Alt+E	Play preroll and selection
Home	Move the current-time indicator to the beginning of the waveform or session
End	Move the current-time indicator to the end of the waveform or session
Page Up	Move the current-time indicator one page to the left
Page Down	Move the current-time indicator one page to the right
Left Arrow	Move the current-time indicator to the left
Right Arrow	Move the current-time indicator to the right

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ug.book Page 264 Tuesday, March 16, 2004 1:29 PM

Keys for selecting ranges, channels, and tracks

Up Arrow	Select the left channel or next higher track
Down Arrow	Select the right channel or next lower track
Shift+Home	Extend the selection to the beginning of the waveform or session
Shift+End	Extend the selection to the end of the waveform or session
Shift+Page Up	Extend the selection one page to the left
Shift+Page Down	Extend the selection one page to the right
Shift+Left Arrow	Extend the selection to the left
Shift+Right Arrow	Extend the selection to the right
Ctrl+Shift+A	Select the current page
[	Move left side of the selection inward during playback
]	Move right side of the selection inward during playback

Keys for copying waveforms

Ctrl+Insert	Copy the waveform or selection to the clipboard
Shift+Insert	Paste the clipboard's contents into the waveform display or session display
Ctrl+M	Insert the waveform into the session display
Ctrl+Shift+N	Paste the contents of the active clipboard to a new waveform

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ug.book Page 265 Tuesday, March 16, 2004 1:29 PM

Keys for editing clips

Ctrl+Up Arrow	Select the previous clip in the currently selected track
Ctrl+Down Arrow	Select the next clip in the currently selected track
Alt+Left Arrow	Nudge the selected clip to the left
Alt+Right Arrow	Nudge the selected clip to the right
Ctrl+Shift+Up Arrow	Clip color (next)
Ctrl+Shift+Down Arrow	Clip color (previous)

Keys for repeating commands

F2	Repeat the last command (its dialog box appears)
F3	Repeat the last command (no dialog box appears)

Keys for using markers

F8	Add a cue or cue range
Shift+F8	Add a CD track marker
Ctrl+F8	Add a CD index marker
1	Mark Intro Time
2	Mark Sec Tone

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Keys for scrolling waveforms and sessions

Ctrl+Home	Scroll to the beginning
Ctrl+End	Scroll to the end
Ctrl+Page Up	Scroll one page to the left
Ctrl+Page Down	Scroll one page to the right
Ctrl+Left Arrow	Scroll to the left
Ctrl+Right Arrow	Scroll to the right

Keys for viewing windows

F12	Toggle between Multitrack View and Edit View
Alt+1	Set focus to the main display
Alt+Page Up	Activate the previous floating window
Alt+Page Down	Activate the next floating window
Alt+/	Flash the window that's currently in focus

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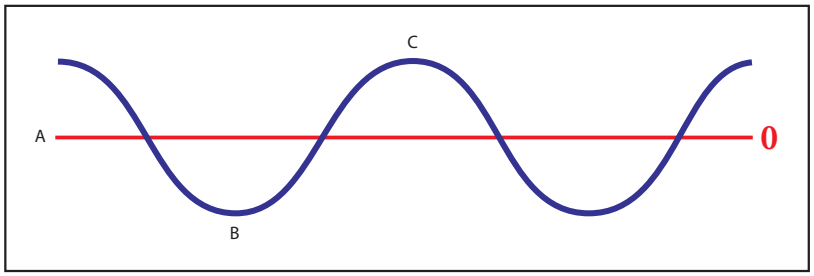
Appendix B: Digital Audio Primer

Understanding the fundamentals of sound is the first step in learning about digital audio. In this primer, we'll introduce the basics of sound so you can work more effectively with Adobe Audition and the rest of your digital audio or video toolkit.

Sound fundamentals

Sound is created by vibrations, such as those produced by a guitar string, vocal cords, or a speaker cone. These vibrations move the air molecules near them, forcing molecules together, and as a result raising the air pressure slightly. The air molecules that are under pressure then push on the air molecules surrounding them, which push on the next set of air molecules, and so forth, causing a wave of high pressure to move through the air. As high pressure areas move through the air, they leave low pressure areas behind them. When these pressure lows and highs—or waves—reach us, they vibrate the receptors in our ears, and we hear the vibrations as sound.

When you see a visual waveform that represents audio, that waveform represents these pressure waves. The zero line in the waveform is the pressure of air at rest. When the line swings up, it represents higher pressure, and when it swings low, it represents lower pressure. This waveform is the equivalent of the pressure waves in the air.



A sound wave represented as a visual waveform
A. *Zero line* ***B.*** *Low pressure area* ***C.*** *High pressure area*

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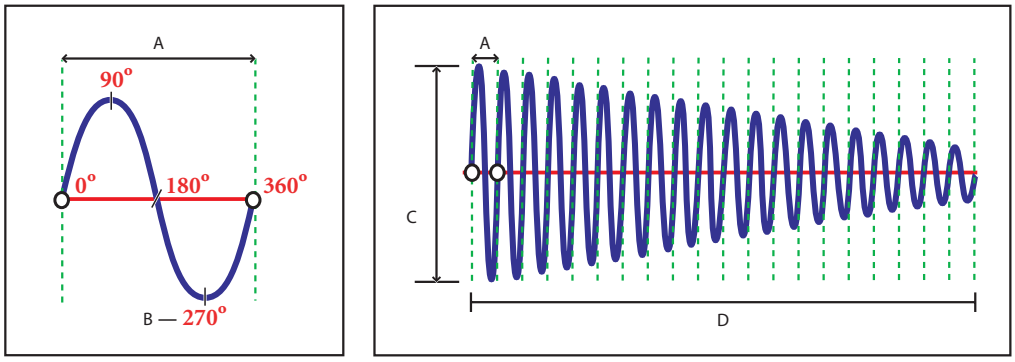
Waveforms

Amplitude reflects the change in pressure from the peak of the waveform to the trough. *Cycle* describes the amount of time it takes a waveform to return to the same amplitude level. *Frequency* describes the number of cycles per second, where one Hertz (Hz) equals one cycle per second. That is, a waveform at 1000 Hz goes through 1000 cycles every second. *Phase* measures how far through a cycle a waveform is. There are 360 degrees in a single cycle; if you start measuring at the zero line, a cycle reaches 90 degrees at the peak, 180 degrees when it crosses the zero line, 270 degrees at the trough, and 360 degrees when it completes at zero. *Wavelength* is the distance, measured in units such as inches or centimeters, between two points with the same degree of phase.



A single cycle at left; a 20 Hz waveform at right
A. *Wavelength* ***B.*** *Degree of phase* ***C.*** *Amplitude* ***D.*** *One second*

When two or more sound waves meet, their amplitudes add to and subtract from each other. If the peaks and troughs of the two waveforms line up, they are said to be in phase. In this case, each peak adds to the peak in the other waveform, and each trough subtracts from the other trough, resulting in a waveform that has higher amplitude than either individual waveform.

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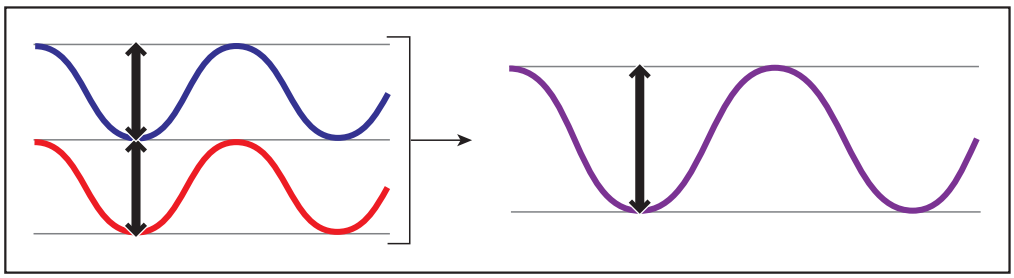
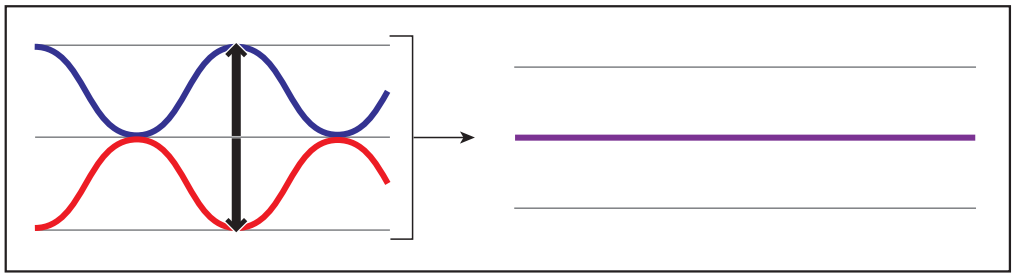


In-phase waves reinforce each other.

Sometimes the peaks of one waveform match up with the troughs of another. The peaks and troughs will cancel each other out, resulting in no waveform at all. Such waveforms are said to be 180 degrees out of phase.



Out-of-phase waves cancel each other out.

In all other cases, waves are out of phase by some amount. This results in a waveform that is more complex than either of the original waveforms; continuing to add waves makes a more and more complicated waveform. Keep in mind, however, that a single instrument can create extremely complex waves because of the unique structure of the instrument; a violin and a trumpet sound different even when playing the same note. When you see music, voice, noise, and other complicated sounds represented by a waveform, you see all the waveforms from each sound added together.

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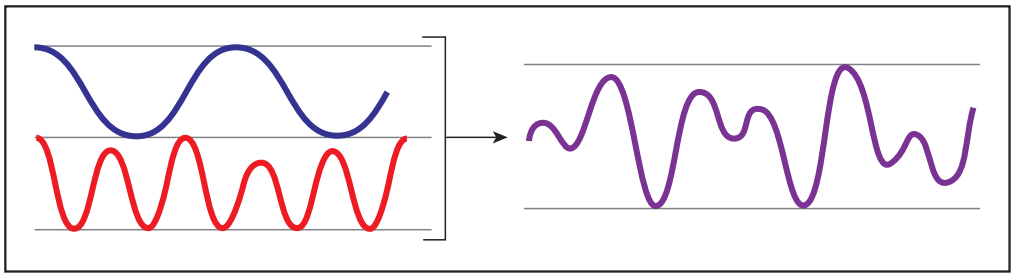

Two simple waves combine to create a complex wave.

Analog audio

A microphone works by converting the pressure waves of sound into changes in voltage on a wire. These changes in voltage match the pressure waves of the original sound: high pressure is represented by positive voltage, and low pressure is represented by negative voltage. Voltages travel down the microphone wire and can be recorded onto tape as changes in magnetic strength or onto vinyl records as changes in amplitude in the groove. A speaker works like a microphone in reverse, taking the voltage signals from a microphone or recording and vibrating to re-create the pressure wave.

Digital audio

Unlike analog storage media such as magnetic tape and vinyl records, computers store audio information digitally as a series of zeroes and ones. In digital storage, the original waveform is broken up into individual samples. This process is typically known as *digitizing* or *sampling* the audio, but it is sometimes called *analog-to-digital conversion*. The sampling rate defines how often a sample is taken. For example, CD-quality sound has 44,100 samples for each second of a waveform.

Sampling rate

The sampling rate determines the frequency range of an audio file. The higher the sampling rate, the closer the shape of the digital waveform will be to that of the original analog waveform. Low sampling rates limit the range of frequencies that can be recorded, which can result in a recording that poorly represents the original sound.

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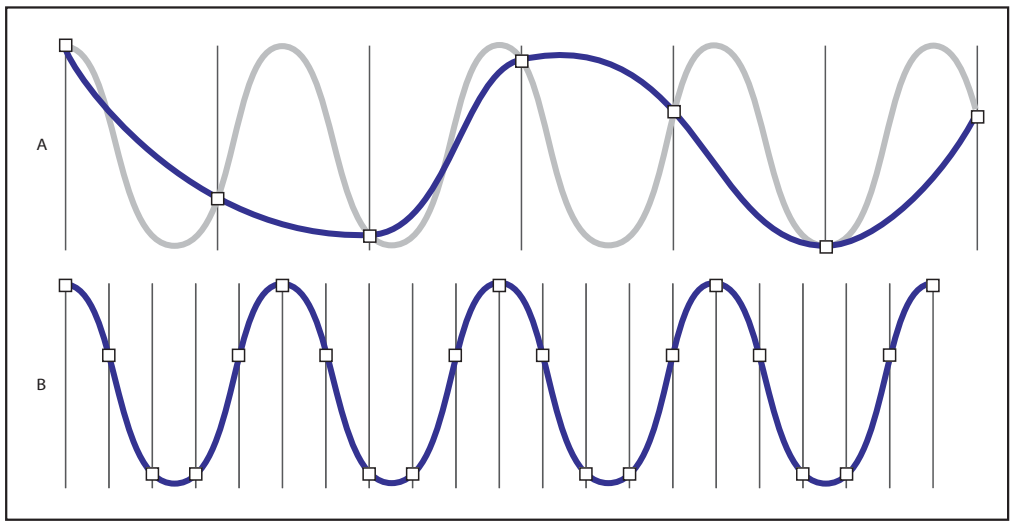

Two sample rates
A. *Low sample rate that distorts the original sound wave.*
B*. High sample rate that perfectly reproduces the original sound wave.*

To reproduce a given frequency, the sampling rate must be at least twice that frequency. For example, if the audio contains audible frequencies as high as 8000 Hz, you need a sample rate of 16,000 samples per second to represent this audio accurately in digital form. This calculation comes from the Nyquist Theorem, and the highest frequency that can be reproduced by a given sample rate is known as the Nyquist Frequency. CDs have a sample rate of 44,100 samples per second that allows sampling up to 22,050 Hz, which is higher than the limit of human hearing, 20,000 Hz.

Bit depth

Just as the sample rate determines the frequency resolution, the bit depth determines the amplitude resolution. A bit is a computer term meaning a single number that can have a value of either zero or one. A single bit can represent two states, such as on and off. Two bits together can represent four different states: zero/zero, one/zero, zero/one, or one/one. Each additional bit doubles the number of states that can be represented, so a third bit can represent eight states, a fourth 16, and so on.

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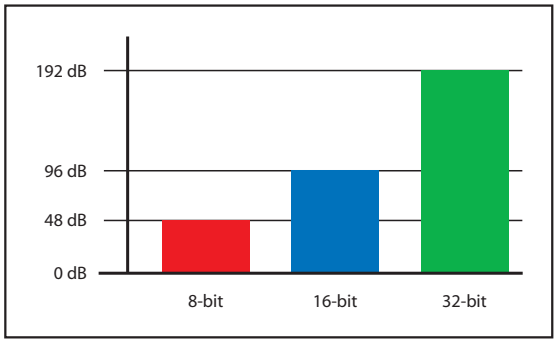
Amplitude resolution is just as important as frequency resolution. Higher bit depth means greater dynamic range, a lower noise floor, and higher fidelity. When a waveform is sampled, each sample is assigned the amplitude value closest to the original analog wave. With a resolution of two bits, each sample can have one of only four possible amplitude positions. With three-bit resolution, each sample has eight possible amplitude values. CD-quality sound is 16-bit, which means that each sample has 65,536 possible amplitude values. DVD-quality sound is 24-bit, which means that each sample has 16,777,216 possible amplitude values.



Higher bit depths provide greater dynamic range.

Where Adobe Audition fits into the process

When you record audio on your computer, Adobe Audition tells the sound card to start the recording process and specifies what sampling rate and bit depth to use. The sound card determines the supported sample rates and bit depths. Most cards can record and play at CD-quality settings, but many also support other settings (for example, a 48 kHz sample rate, which is common in film and video post-production). Your sound card probably has both Line In and Microphone In ports through which it can accept analog signals. The sound card samples the audio at the specified sample rate and assigns each sample an amplitude value. Adobe Audition stores each sample in sequence until you stop recording. Once you've recorded the audio, you can use Adobe Audition to edit the audio or save it to disk as a file.

When you play a file in Adobe Audition, the process happens in reverse. Adobe Audition tells the sound card that it will play a file and sends a series of digital samples to the card. The sound card reconstructs the original waveform and sends it as an analog signal through the Line Out port to your speakers.

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An audio file on your hard drive, such as a WAV file, consists of a small header indicating sample rate and bit depth, and then a long series of numbers, one for each sample. These files can be very large. For example, at 44,100 samples per second and 16 bits per sample, a file includes 705,600 bits per second. This equals 86 kilobytes per second and more than 5 megabytes per minute. Stereo sound has two channels, so CD-quality sound requires a little more than 10 megabytes per minute.

Introducing MIDI

In contrast to a digital audio file, a MIDI file might be as small as 10 kilobytes per minute, so you can store up to one hundred minutes of MIDI per megabyte. MIDI and digital audio are fundamentally different: digital audio is a digital representation of a sound wave, MIDI is a language of instructions for musical instruments. A digital audio file seeks to exactly represent an audio event just like a tape recorder, whether it's a musical performance, a person talking, or any other sound. MIDI, on the other hand, is more like sheet music. It acts as instructions for the re-creation of a musical selection. These MIDI instructions, however, cannot reproduce highly complex sounds, such as the human voice.

MIDI files record information such as the note to be played, the instrument to play the note on, the pan and volume of that particular note, and so on. When a MIDI file is played back, the sound card takes this information and uses its synthesizer to re-create the note on the right instrument. Because every synthesizer sounds different, the MIDI file will sound different when played through different sound cards. MIDI support in Adobe Audition is limited to playback of MIDI files.

Conclusion

To summarize, the process of sampling or digitizing audio starts with a pressure wave in the air. A microphone converts this pressure wave into voltage variations. An analog-to-digital converter, found in devices such as sound cards, samples the signal at the sample rate and bit depth you choose. Once the sound has been transformed into digital information, Adobe Audition can record, edit, process, mix, and save your digital audio files. The possibilities for manipulation of digital audio within Adobe Audition are limited only by your imagination.

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Appendix C: Glossary

A

ACM (Audio Compression Manager) A Microsoft technology that enables Windows applications to compress and decompress files in a variety of formats, such as DSP Group TrueSpeech and GSM 6.10. Some ACM formats install with Windows, while others install with software applications.

ADPCM (Adaptive Differential Pulse Code Modulation) An audio compression scheme that compresses sound files from 16 bits to 4 bits, yielding a 4:1 compression ratio. There are many varieties of ADPCM, such as the IMA (Interactive Multimedia Association) DVI standard, and versions from Microsoft, Dialogic, and others.

ActiveMovie See "DirectX" on page 279.

Adapter A cable, plug, or jack that enables you to connect two audio or video devices together.

ADAT A digital 8-track tape deck manufactured by Alesis Corporation that is very popular in recording studios.

Aliasing Noise that occurs when a high frequency sound exceeds the Nyquist Frequency for a given sample rate. (See "Nyquist Frequency" on page 284.) Most analog-to-digital converters prevent aliasing by filtering out sounds above the Nyquist Frequency.

Amplitude Amplitude represents the loudness of an audio signal. A waveform's amplitude is measured by its distance from the center line, which represents an amplitude of 0. There are different standards for measuring amplitude, but the decibel (dB) is the most common. (See "Decibel (dB)" on page 279.)

Analog recording Traditional audio recording with devices such as magnetic tape machines and vinyl records. Analog audio recording consists of a continuous curve, as opposed to digital recording, which consists of discrete samples.

ASCII text data You can represent audio data in this standard text format (.txt), with each sample separated by a carriage return, and channels separated by a tab character. Before the audio data, you can add a header with a format of `Keyword:Value`, with the keywords `Samples`, `BitsPerSample`, `Channels`, `SampleRate`, and `Normalized`. (The values for `Normalized` are `True` or `False`.) If no header exists, the data is assumed to be 16-bit signed decimal integers.

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Attack The first part of the sound that you hear. Some sounds (like pianos and drums) have a very fast attack; the loudest portion of the sound occurs very quickly. A sound with a slow attack rate (such as a soft string section) slowly increases in volume.

Attenuate To reduce volume or signal level.

Audio file format The method used to store audio data on disk, chosen in Save dialog boxes. Adobe Audition supports many file formats, and each supports a variety of properties such as sample rate and compression. Some file formats may not be compatible with other platforms. On the Windows platform, Windows PCM (.wav) is the most common format.

Audition loop See "Audition Loop (.cel)" on page 234.

Automation The process of recording volume and pan changes during a mix, and perfectly reproducing those changes every time a mix plays. In hardware mixers that support automation, volume and pan controls record timing information and physically move during playback. In Adobe Audition, you automate mixes with visual envelopes. (See "Envelopes" on page 280.)

B

Background mixing The process that Adobe Audition uses to mix audio for playback in Multitrack View. Background mixing occurs behind the scenes, reflecting changes to a session, such as a moved or deleted clip, a volume change, or a newly recorded track. The progress of background mixing is displayed by the Mix Gauge. (See "Mix Gauge" on page 283.)

Band pass filter A filter that allows some audio frequencies to pass through unchanged.

Basic cue One of four types of Adobe Audition cues. Basic cues mark important sections of a waveform for later reference (for example, to identify an editing point). These cues also specify stop and start positions for the Play List. (See "Play List" on page 285.)

Beat cue One of four types of Adobe Audition cues. Beat cues function like basic cues, but they specifically identify musical beats.

Beats per minute (bpm) Musical tempo, which is defined by the number of beats that occur every 60 seconds.

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Bit Part of the numbering system used in digital data. Bits are combined in groups to form digital words, which represent the changing amplitude values of an analog signal. Bit resolution describes the number of bits used in each word, determining the number of possible amplitude values. Therefore, higher bit resolutions produce higher dynamic range.

Bit resolution (or bit depth) The number of bits used to represent audio amplitude. 8-bit resolution provides a maximum of 256 unique amplitude levels, producing a 48 dB dynamic range; 16-bit resolution provides 65,536 unique amplitude levels, producing a 96 dB dynamic range. Compact disc players have 16-bit resolution, but some sound cards support resolutions higher than 16-bit. Adobe Audition supports up to 32-bit resolution. For the best audio quality, remain at the 32-bit level while transforming audio in Adobe Audition, and then convert to a lower resolution for output.

Brown noise Brown noise has a spectral frequency of 1/f^2, so it emphasizes low-frequency components, resulting in thunder- and waterfall-like sounds. Brown noise follows a Brownian motion curve, in which each sample in a waveform contains a mixture of predefined and random frequency components.

Bus In hardware mixers, a channel that lets you combine several other channels and output them together. In Adobe Audition's Multitrack View, you can similarly use software buses to combine several tracks.

Burn To write to a CD-R or CD-RW disc.

C

CD-R A recordable compact disc that you can write to only once. These discs typically hold 650 MB of data, which equals 74 minutes of stereo audio. *CD-R* sometimes refers to the computer drives that burn CD-R discs.

CD-RW A rewritable compact disc. These discs typically hold 650 MB of data, which equals 74 minutes of stereo audio. Unlike a CD-R, however, a CD-RW disc can be erased and written to again.

Chorus A delay effect that simulates several voices by adding multiple short delays with a medium amount of depth and a small amount of feedback.

Click track An audio track comprised of clicks that occur on the beat, like a metronome. Click tracks are often used at the beginning of a session to provide timing information for musicians and then removed from the session before mixing down.

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ug.book Page 278 Tuesday, March 16, 2004 1:29 PM

Clip A visual representation of individual audio, video, or MIDI files in Adobe Audition's Multitrack View.

Clipping In digital audio, distortion that occurs when the amplitude of a signal exceeds the maximum level for the current bit resolution (for example, 256 in 8-bit audio). Visually, clipped audio produces broad flat areas at the top of a waveform. If you experience clipping, lower the recording input or the source output levels.

CODEC (Compressor/Decompressor) An abbreviation often used to describe multimedia compression schemes used by ACM, MPEG, QuickTime, AVI, and the combined A-D-D-A modules on some sound cards.

Compressor Reduces dynamic range by lowering amplitude when an audio signal rises above a specified threshold. For example, compressors can be used to eliminate variations in the level of an electric bass, providing an even, solid bass line. Compressors can also compensate for variations in level produced by a vocalist who moves frequently or has an erratic volume.

Crossfade A fade from one audio track to another.

Crosstalk Undesired leakage of audio from one track to another, a common problem with analog tape. Crosstalk is impossible in Adobe Audition because each track is stored as a separate digital audio file.

Cue List A list of time locations defined in an audio file. A cue can be either a point that specifies a time position or a range that specifies a selection. In Adobe Audition, you can define and save an unlimited number of cues for later recall or for assembly in the Play List window. (See "Play List" on page 285.)

D

DAC (Digital-to-Analog Converter) The hardware responsible for converting a digital audio or video signal into an analog signal that you can play through amplifiers and speakers.

DAT (Digital Audio Tape) A standard two-track digital audio tape format. DAT tapes are sampled at 16 and 24 bits, and 32,000, 44,100, and 48,000 samples per second. (The latter is often described as *DAT quality*).

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ug.book Page 279 Tuesday, March 16, 2004 1:29 PM

DC offset Some sound cards record with a slight DC offset, in which direct current is introduced into the signal, causing the center of the waveform to be offset from the zero point (the center line in the waveform display). DC offset can cause a click or pop at the beginning and end of a file. To compensate for DC Offset, use the DC Bias Adjust setting provided by the Amplify command.

Decibel (dB) In audio, the decibel (dB) is a logarithmic unit of measurement used for amplitude.

dBFS Decibels below full scale in digital audio. 0 dBFS is the maximum possible amplitude value (for example, 256 for 8-bit audio). A given dBFS value does not directly correspond to the original sound pressure level measured in acoustic dB.

Delay A time-shifted signal that you can mix with the original, nondelayed signal to provide a fuller sound or create echo effects. Adobe Audition offers a variety of delay effects such as Reverb, Chorus, and Echo.

Destructive editing Editing (such as cutting and pasting, or effects processing) that changes the original audio data. For example, in destructive editing, a change in audio volume alters the amplitude of the original wave file. In Adobe Audition, Edit View is a destructive editing environment; however, edits do not permanently change audio until you save a file.

Devices Wave and MIDI devices that send data into and out of the computer. In Adobe Audition, wave devices are sound card inputs and outputs used for recording and playback of audio; MIDI devices are hardware interfaces used to send performance and synchronization information to Adobe Audition and other MIDI-enabled programs and hardware. You can configure both device types in the Device Properties dialog box.

Digital Signal Processing (DSP) The process of transforming a digital audio signal by using complex algorithms. Examples include filtering with equalizers, and effects processing with reverbs and delays.

DirectX A development platform designed by Microsoft that provides an open standard for audio plug-ins. Plug-ins based on this standard can be used by any application that supports DirectX, such as Adobe Audition.

Dither Dithering adds small amounts of noise to a digital signal so that very quiet audio remains audible when you convert from a high bit resolution to a lower one (for example, when converting from 32-bit to 16-bit). Without dithering, quiet audio passages such as long reverb tails may be abruptly truncated.

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ug.book Page 280 Tuesday, March 16, 2004 1:29 PM

Dry Used to describe an audio signal without any signal processing such as reverb; the opposite of Wet.

DSP See "Digital Signal Processing (DSP)" on page 279.

DVD (Digital Video Disc) A storage medium similar to a compact disc (CD), but with much higher bandwidth and storage capabilities. Audio stored in DVD movies is generally 96 kHz/24-bit.

E

Echo A distinct repetition of a sound, caused by the sound reflecting off a surface. Adobe Audition offers two echo effects, Echo and Echo Chamber.

8-bit Signed See "8-bit signed (.sam)" on page 232.

Envelopes To automate mixes in Multitrack View, Adobe Audition uses envelopes, which are drawn directly on clips. Envelopes visually indicate the pan, volume, wet/dry, and effects parameter settings at any point in a track. For example, when a volume envelope is at the top of an audio clip, the audio is at full volume; when the envelope is at the bottom, the audio is at zero volume.

Equalization (EQ) The process of increasing or decreasing the amplitude of specific audio frequencies relative to the amplitude of other audio frequencies.

Expander Increases dynamic range by lowering amplitude when an audio signal falls below a specified threshold (the opposite of a compressor). For example, an expander can be used to lower the level of background noise that becomes audible when a musician stops playing.

F

Fast Fourier Transform (FFT) An algorithm based on Fourier Theory that Adobe Audition uses for filtering, Spectral View, and Frequency Analysis features. Fourier Theory states that any waveform consists of an infinite sum of sin and cos functions, allowing frequency and amplitude to be quickly analyzed. Higher FFT sizes create more precise results but take longer to process.

Flange An audio effect caused by mixing a varying, short delay in roughly equal proportion to the original signal.

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ug.book Page 281 Tuesday, March 16, 2004 1:29 PM

Flushing The process Adobe Audition performs when it copies the audio data from a waveform file to Adobe Audition's temp folder so that the original file can be closed. This allows the file to be renamed, deleted, or opened exclusively by another application. Flushing sometimes occurs when a modified waveform is saved on top of its original file.

Frequency Measured in Hertz (Hz), cycles per second, frequency describes the rate at which a sound wave vibrates. A cycle consists of movement from a starting point (0) through both positive and negative amplitudes, eventually returning to the starting point. A sound's frequency determines its pitch: high frequency equals high pitch, and low frequency equals low pitch.

FX An abbreviation for *effects*.

H

Hertz (Hz) Cycles per second. A unit of measurement that describes the frequency of a sound. (See "Frequency" on page 281.)

I

Index cue One of four types of Adobe Audition cues. Index cues become index markers in a CD track. If a CD player is configured to display remaining time, it displays the time before track markers and index markers. Note, however, that not all CD players support index markers.

Impulse A data file that the Convolution effect uses to modify samples. Impulses function like amplitude maps. For example, if you apply an impulse of a single full-volume sample, the original audio data will be unchanged. Should the impulse be at half volume, however, the original audio data will be reduced to half volume. If several such impulses occur over time, each with descending amplitude, the original audio data will gradually and rhythmically become lower in volume.

Interpolate To estimate the values of data points between known data points. Interpolation is used when new data must be generated to fill in areas where values are unknown.

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ug.book Page 282 Tuesday, March 16, 2004 1:29 PM

L

Level Meters Adobe Audition's Level Meters are found by default along the bottom of the application window, and they are used to monitor the volume of incoming and outgoing signals. The red clip indicator to the right of the meters will light up and remain lit when levels exceed the maximum of 0 dB. Clicking the clipping indicator resets it. The top meter represents the left channel, and the bottom meter represents the right.

Limiter A signal processor that limits input signals that exceed a specified threshold level. Above the threshold, the output level remains constant even if the input increases in volume.

Loop An audio file that contains tempo and pitch information, allowing it to match the tempo and pitch of other loops in a multitrack session. You can repeat a loop-enabled clip infinitely by simply dragging its bottom right corner.

M

Mastering The process of finalizing audio for a specific medium, such as the Web or audio CD. Mastering consists of several processing phases, with equalization and compression phases being the most essential. You can master audio files either individually or in groups. (Collectively mastering groups of files is particularly important if the destination medium is audio CD.)

MIDI Musical Instrument Digital Interface, a way of communicating performance information from one piece of software or hardware to another. MIDI can simply relay musical notes, or it can transmit detailed information about timing, synthesizer patches, and such. Windows transmits MIDI information internally between applications; to transmit MIDI information to and from your computer and external devices such as MIDI keyboards, you must use a hardware MIDI interface (for example, the MIDI In port of a sound card).

MIDI Timecode (MTC) A method of sending timing information between MIDI-capable devices. For example, you can convert SMPTE timecode to MTC to synchronize Adobe Audition's transport controls with a video or audio tape deck.

MIDI Trigger An Adobe Audition shortcut triggered by a MIDI event, such as Note On. You can send MIDI events to any device capable of issuing a MIDI command, such as MIDI keyboards and sequencers.

Millisecond (ms) One thousandth of a second. (There are 1000 milliseconds in a second.)

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ug.book Page 283 Tuesday, March 16, 2004 1:29 PM

Miniplug A common name for 1/8-inch plugs and jacks, sometimes known as *minijacks*. On the most common sound cards, miniplug jacks provide analog audio inputs and outputs.

Mixdown The process of combining the output of several tracks in Multitrack View to create a new stereo waveform. When you mix down, track properties such as Volume and Pan are reflected in the resulting waveform, so mixdown is typically performed when you're happy with the sound of a session. A mixdown can also produce submixes of selected tracks. For example, you could create a submix of multiple drum tracks and place it on a single, open track, cleaning up the Multitrack View workspace.

Mixing The process of combining multiple audio sources or tracks together for output as a single source. Output is generally in the form of a stereo pair of channels, though mixes may be directed to any number of channels for output (for example, one channel for monophonic output, or 6 channels for surround-sound output).

Mix Gauge Found below the track controls area in Multitrack View, the Mix Gauge indicates the progress of background mixing. Whenever you edit a session, the Mix Gauge becomes blank and then gradually fills as the mix is reprocessed, turning brighter in color when background mixing is complete. You don't need to wait for the Mix Gauge to finish before playing a session.

Mono A monophonic signal, which contains only one sound source.

N

Noise gate A special type of expander that reduces or eliminates noise by greatly lowering signal levels that fall below a specified threshold. Noise gates are often configured to totally eliminate background noise during musical pauses. You can also use these gates to silence pauses in speech.

Noise shaping A technique that shifts the frequency of dithering noise to minimize its audibility.

Nondestructive editing Nondestructive edits don't alter a sound file on disk in any way. For example, nondestructive volume changes do not alter the amplitude of a waveform, but instead simply instruct an audio application to play the waveform at higher volume. In Adobe Audition, Multitrack View is a nondestructive editing environment.

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ug.book Page 284 Tuesday, March 16, 2004 1:29 PM

Normalize To adjust the highest peak of a waveform to a certain percentage relative to the digital maximum, 0 dBFS, thereby raising or lowering all other peaks accordingly. Typically, audio is normalized to 100% to achieve maximum volume, but Adobe Audition lets you normalize to any percentage.

Nyquist Frequency Also called Nyquist Rate, this frequency equals half the current sample rate and determines the highest reproducible audio frequency for that sample rate. For example, audio CDs use a sampling rate of 44,100 Hz because the resulting Nyquist Frequency is 22,050 Hz— just above the limit of human hearing, 20,000 Hz. Likewise, to reproduce a signal with an 11,000 Hz frequency range, you must use a sample rate of at least 22,000 Hz. To avoid aliasing distortion, nearly all analog-to-digital converters filter out frequencies that exceed the Nyquist Frequency before the analog-to-digital conversion process. For the best audio quality, record and edit at higher sample rates and then convert down if needed.

O

Offline editing See "Destructive editing" on page 279.

Order A value that determines the slope of an audio filter. First-order filters attenuate an additional 6 dB per octave, second-order filters attenuate 12 dB, third-order filters 18 dB, and so on.

P

PCM (Pulse Code Modulation) PCM is the standard method used to digitally encode audio and is the basic, uncompressed data format used in file formats such as WAV and AIFF.

Peak files Cache files with the extension .pk that enable Adobe Audition to open, save, and redraw audio files more quickly. You can safely delete peak files or deselect the Save Peak Cache Files option in the Settings dialog box. However, keep in mind that without peak files, larger audio files will reopen more slowly.

Phase The position of a sound wave relative to other sound waves. As a sound wave travels through the air, it compresses and expands air molecules in peaks and troughs, much like an ocean wave. In the waveform display, peaks appear above the center line, troughs appear below. If two channels of a stereo waveform are exactly opposite in phase, they will cancel each other out. More common, however, are slightly out-of-phase waves, which have misaligned peaks and troughs, resulting in duller sound.

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ug.book Page 285 Tuesday, March 16, 2004 1:29 PM

Pink noise Noise with a spectral frequency of 1/f, producing the most natural-sounding generated noise. By equalizing pink noise, you can simulate rainfall, waterfalls, wind, a rushing river, and other natural sounds. On the audio spectrum, pink noise falls exactly between brown and white noise.

Play List An arrangement of Cue List entries that you can play in any order and loop a specified number of times in nondestructive fashion. Adobe Audition saves Play Lists in the header of WAV files.

Plug-in A software component that you can add to another piece of software to increase its functionality. Adobe Audition supports third-party VST and DirectX audio plug-ins, which seamlessly integrate into Adobe Audition's interface.

Preset Most dialog boxes in Adobe Audition support presets, which are settings saved under a particular name for later recall. Dialog boxes that support presets have a Preset list where you can click a preset to recall its settings, and Add and Del buttons for creating and deleting presets.

Preview Many dialog boxes in Adobe Audition offer real-time Preview buttons, letting you monitor setting changes as you make them. The preview quality depends upon your system's performance.

Punch in A recording method used to insert a new recording into a specific region of an existing waveform, usually to replace an undesirable section. Adobe Audition supports punch-in recording in Multitrack View and allows for multiple takes; you can repeatedly record over the original material and afterward choose the best performance.

Q

Quantization A process that occurs when an analog waveform is converted to digital data and becomes a series of samples. Quantization noise is introduced as some samples are shifted to quantization levels allowed by the current bit resolution. This noise is highest at low bit resolutions, where it can particularly affect low amplitude sounds.

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ug.book Page 286 Tuesday, March 16, 2004 1:29 PM

R

RCA cable Sometimes called a phono cable, RCA cables have RCA plugs or jacks at either end and are normally used to connect stereo system components, such as receivers, CD players, and cassette decks.

Real time In computer-based audio, *real time* refers to functions that react immediately to user input and transform audio nondestructively. (Note, however, that system speed ultimately determines processing time.) Adobe Audition provides real-time mixing and effects in Multitrack View, and real-time effects previews in Edit View.

Referenced clip In Multitrack View, a referenced clip shares a source file with other clips. For example, if a drum hit occurs 30 times in a session, you can conserve disk space by using 30 referenced clips of the same source file. Because referenced clips represent the same file, any alteration to a referenced clip (like a cut or transform) affects all instances in a session. By contrast, unique clip copies create a separate sound file on disk, consuming more disk space, but allowing for separate editing.

Resample To convert a sound file to a different sample rate.

Reverb The reverberant sound produced by an acoustic space, such as a room or concert hall. Reverb consists of dense, discrete echoes that arrive at the ear so rapidly that the ear can't separate them. Adobe Audition offers four reverb effects: Quick Verb, Studio Reverb, Reverb, and Full Reverb.

Rip The process of digitally extracting audio from a compact disc and turning it into a waveform. Most newer CD-ROM, CD-R, and CD-RW drives support digital audio extraction.

RMS (Root-mean-square) A mathematical formula used to determine the average amplitude of an audio selection. RMS amplitude reflects perceived loudness better than peak amplitude.

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ug.book Page 287 Tuesday, March 16, 2004 1:29 PM

S

S/N ratio Signal-to-noise ratio describes the difference between the highest signal level before distortion and the average level of the noise floor. In most analog systems, such as microphone preamps, the S/N ratio is around 92 dB.

Sample A digital snapshot of an audio waveform at a particular point in time. In digital audio, a series of numeric samples reproduces an entire waveform, with higher sample rates producing increased frequency response. (Note that musical samplers use the term *sample* to describe a digital recording, rather than a digital snapshot.)

Sample rate The number of samples per second. Higher sample rates produce increased frequency response but require more disk space. To reproduce a given audio frequency, the sample rate must be at least double that frequency. (See "Nyquist Frequency" on page 284.)

Sampler A musical device that records and plays digital sounds (known as *samples* in this context) and lets you edit and store those sounds.

Sequencer A programmable electronic device that can record and play a sequence of musical events, such as samples, pitches, and rests. Most modern sequencers are MIDI-based. (See "MIDI" on page 282.)

Session A multitrack project in Adobe Audition. Session files are stored with the extension .ses and contain details such as mixing and effects settings. Session files don't contain audio data; instead they contain pathnames pointing to the sound files used in the session.

64-bit Doubles See "64-bit doubles (RAW) (.dbl)" on page 232.

SMPTE timecode (Society of Motion Picture and Television Engineers timecode) A timing reference used to synchronize two devices. SMPTE timecode is divided into hours, minutes, seconds, and frames.

Sound card A hardware device that lets your computer play and record audio.

Sound wave A wave of air molecules. Humans can hear sound waves with frequencies of 20 to 20,000 Hz.

Stereo A signal with a left and right channel, allowing for spatial placement of sounds.

Stripe To copy SMPTE timecode to a single track of a multitrack tape so remaining tracks can be synchronized with other devices.

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ug.book Page 288 Tuesday, March 16, 2004 1:29 PM

T

Tempo The rhythmic speed of music, normally measured in bpm. (See "Beats per minute (bpm)" on page 276.)

Timecode An audio or digital signal that synchronizes time between multiple devices. The most common forms are SMPTE and MIDI timecode.

Track A container for one or more clips in Multitrack View. Each track has independent settings for volume, pan, EQ, effects, and input and output. Each session can have up to 128 tracks.

Track cue One of four types of Adobe Audition cues. Track cues indicate start points for CD tracks.

Track controls The area of Multitrack View that controls each track, with independent settings for volume, pan, EQ, effects, and input and output.

TXT See "ASCII text data" on page 275.

U

Unity gain An amplification level that precisely corresponds to the input signal level, without amplifying or lowering it. (Note that audio hardware operates at two line levels: –10 dBV for consumer equipment, and +4 dBu for professional. If these two hardware types are connected, unity gain will result in a lowered input for consumer equipment, and a raised input for professional.)

W

Wave file Any audio file format that contains primarily sound wave data. Wave files can be in formats such as WAV, AU, AIF, or mp3.

Waveform A term that describes the visual representation of an audio signal, displayed as amplitude across time in Adobe Audition. (In acoustics, *waveform* refers to a sound wave of a specific frequency.)

Waveform clip A visual representation of a wave file or related image in Multitrack View. Edits of these clips are nondestructive.

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ug.book Page 289 Tuesday, March 16, 2004 1:29 PM

Waveform display The area of Edit View in which you view and edit audio data. By default this audio material appears as a waveform, but you can view it in spectral form by choosing View > Spectral View.

Wet Used to describe an audio signal that includes signal processing such as reverb; the opposite of Dry.

White noise White noise has a spectral frequency of 1, so equal proportions of all frequencies are present. Because more individual frequencies exist in the upper ranges of human hearing, white noise sounds very hissy. Adobe Audition generates white noise by choosing random values for each sample.

Z

Zero crossing A point in time where a waveform crosses the zero amplitude line. To make edits sound smoother, place them at zero-crossing points, thus avoiding abrupt changes in amplitude that cause pops and clicks.

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ug.book Page 291 Tuesday, March 16, 2004 1:29 PM

Index

A

absorptive surfaces, simulating with Echo effect 148
ACM
  defined 275
  Waveform format 233
adapter, defined 275
ADAT, defined 275
Add/Remove VST Directory command 33
Adjust Boundaries command 170
Adjust Sample Rate command 111
Adobe Acrobat Reader 1
Adobe After Effects, remixing soundtracks from 207
Adobe Audition
  and basic audio workflow 272
Adobe Premiere Pro, remixing soundtracks from 207
ADPCM
  compression 235, 236
  defined 275
Advanced Session Properties dialog box 167
AIFF format 233
A-Law Wave format 232
aliasing, defined 275
Amiga IFF-8SVX format 233
amp overdrive 155
Amplify/Fade effect 134
amplitude
  defined 275
  optimizing 134
analog audio, fundamentals of 270
analog recording, defined 275
Analyze Loudness tab, Group Waveform Normalize dialog box 245
Append To Session command 63
Apple AIFF format 233
ASCII text data
  about 234
  defined 275
ASPI (Advanced SCSI Programming Interface) 46
attack, defined 276
attenuate, defined 276
AU format 238
audio
  analyzing 118
  copying 92
  mastering 117
  muting 103
  restoring 117
  shrinking or stretching 146, 177
Audio Clip Properties window 174
audio digital extraction 64
audio file format, defined 276
audio files
  See also specific format names
  exporting mixdowns to 229
  inserting in sessions 63
  saving 227
audio tracks, about 179
Audition Loop format 234
Auto Click/Pop Eliminator effect 125
Auto-Cue feature 101
automation, defined 276
auto-play 62, 77
AVI format, exporting mixdowns to 230

B

background mixing
  defined 276
  priority level 54
band pass filter, defined 276
Bars And Beats command 205
Bars and Beats time format
  about 71
  tempo and 199
Basic cue
  about 97
  defined 276
Bass Management circuit 220
batch processing
  about 243
  cue list 243
  cue ranges 243
  normalizing files 244
  with scripts 247
Batch Processing command 247
beat cue, defined 276
beats
  finding 90
  marking with cues 98
beats per minute (bpm)
  and tempo 199
  defined 276
Binaural Auto-Panner effect 139

Downloaded from www.Manualslib.com manuals search engine

ug.book Page 292 Tuesday, March 16, 2004 1:29 PM

bit depth
automatically converting files to 32-bit 51
changing 113
defined 277
downsampling for playback 38
fundamentals of 271
bit resolution, defined 277
bit, defined 277
brown noise
about 108
defined 277
buffers
playback, Edit View 46
playback, Multitrack View 54
real-time preview 46
recording, Edit View 46
recording, Multitrack View 54
burn, defined 277
burning CDs
in Adobe Audition 261
tips when using another burning program 228, 230
Bus Mixer 193
Bus Properties dialog box 193
bus, defined 277

C

CBR 225
CBR encoding 237, 239
CCITT formats 232, 233, 238
CD Project View
switching to 11
using 257
CD-R, defined 277
CD-RW, defined 277
CDs
adding text 260, 262
editing source audio 260
extracting tracks from 64
inserting tracks 258
pre-emphasis 261
properties for CD-burning devices 261
properties for tracks 260
recording from 67
removing tracks 259
selecting tracks 259
setting pauses between tracks 260
writing 261
CEL format 234
Center Channel Extractor effect 141
center lines, showing or hiding 51
Channel Mixer effect 140
channels
converting between mono and stereo 112
selecting data in left or right 88
specifying which to edit 92
Check for Hidden Clips command 179
Chorus effect 143
Click/Pop Eliminator effect 125
clip indicators, clearing 81
Clip Restoration effect 126
clipboard
choosing 92
clearing on exit 48
resampling data 53
clipping
displayed in Level Meters 80
showing or hiding clipping boundaries 51
when normalizing groups of files 246
clips
about 168
aligning 170
changing properties 174
copying 173
crossfading 176
editing 170
envelopes for 188
grouping 169
inserting in sessions 63
locking in time 174
mixing down 196
moving 168
muting 174
protecting from recording 174
rejoining 172
removing from a session 179
repeating 173
selecting 168
snapping 170
splitting 172
time stretching 177
video 209
closing files 25, 231
command line, playing audio from 78
commands, choosing 12
consistent volume 244
Constant Bit Rate (CBR) 225
context-sensitive menus 12
control signal, for Vocoder effect 158

Downloaded from www.Manualslib.com manuals search engine

ug.book Page 293 Tuesday, March 16, 2004 1:29 PM

Convert Sample Type command
using to change the bit depth 114
using to change the sample rate 111
using to convert between stereo and mono 112
using to convert sample rate of multiple files 115
converting
audio formats 247
sample rate of session 229
sample type 61
to a different format 227, 229
Convolution effect 154
copy protection for CDs 260
Copy Reference Here command 173
Copy To New command 199
Copy Unique Here command 173
copying audio 92
Creative Labs Audigy 223
Creative Sound Blaster format 235
Crossfade commands 176
crossfades, duration of 56
crossover frequency points 226
Cue List
batch processing 243
defined 278
Cue List command 243
cues
adding silence between 100
adjusting 99
creating 96
deleting 100
inserting in a session 64
merging 99
playing 97
point cues versus range cues 96
renaming 98
saving 228, 230, 231
saving to files 100
selecting 96
setting automatically 101
showing or hiding cue lines 50
snapping to 91
types of 97
current-time indicator, about 68
custom time format 71
custom timecode display 44
cutting audio 93

D

DC bias
removing when recording 39
DC offset
adjusting 81
Default Session commands 163
Delay effect 147
delay units, and Multitap Delay effect 149
delay, adding to audio 147
Delete Silence command 104
Delete This Take command 75
deleting
audio 95
silence 104
Destroy Clips command 179
Destroy Selected Tracks (Remove and Close) command 260
Device Order command 36
Device Properties command 37, 38
devices
CD 261
choosing which devices to use 36
general uses of 36
setting properties for input 38
setting properties for output 37
Dialogic ADPCM format 235
DiamondWare Digitized format 235
digital audio, fundamentals of 270
Digital Theater Systems (DTS) 226
DirectX
plug-ins 32
using with surround sound 214
disk space
freeing up 57
monitoring 22
Display Time Format command 70
display window, navigating in 17
Distortion effect 155
dithering
applying for playback 38
in response to lower bit depths 113
preferences 52
when saving 32-bit data to 16-bit files 53
docking windows 14
Dolby encoder 213
Doppler Shifter effect 142
downsampling
audio during playback 38
quality level 53
DTMF signals, generating 106
DTS 226
DTS encoder 213
DVD, defined 280
DVI/IMA ADPCM format 236
DWD format 235
Dynamic Delay effect 147
Dynamic EQ effect 129
Dynamic Peaks option 81

Downloaded from www.Manualslib.com manuals search engine

ug.book Page 294 Tuesday, March 16, 2004 1:29 PM

dynamic range, displayed in Level Meters 81
Dynamics Processing effect 136

E

echo
  adding to audio 147
  defined 280
Echo Chamber effect 149
Echo effect 148
Edit Favorites command 253
Edit Original functionality, embedding in files 51, 207
Edit Tempo command 199
Edit View
  considerations for using 83
  switching to 11
  using 10
  using effects in 139
editing audio
  See also specific editing tasks
  general considerations 83
  in Multitrack View 170
  undoing and redoing 22
effects
  See also specific effect names
  cancelling 54
  general use of 28
  graph controls 29
  grouping in Effects tab 26
  plug-ins 32
  presets 28
  real-time, about 185
  using in Edit View 139
Effects Rack dialog box 185
Effects tab in Organizer window 26
8-bit signed format 232
empty audio clips, inserting 178
Enable DirectX Effects command 32
Enable Preroll And Postroll Preview command 32
enhancing audio, about 117
Envelope effect 135
Envelope Follower effect 157
envelopes
  defined 280
  for clips 188
equal loudness contour 246
equalization (EQ), defined 280
expander, defined 280
exporting
  mixdowns to audio 229
  mixdowns to video 230
Extract Audio From CD command 65
Extract Audio From CD options 66

F

fading audio files 134
Fast Fourier Transform (FFT), defined 280
fast-forwarding 79
favorites
  about 253
  viewing in Favorites tab 27
FFT Filter effect 130
files
  See also specific format names
  backward compatibility 51
  choosing a format 231
  closing 25
  converting to a different format 227, 229
  creating new 84
  linking to Adobe Premiere Pro or After Effects 51, 207
  monitoring size of 22
  opening 61
  properties, adding 116
  renaming 249
  saving audio to 227
Files tab 24
filtering audio, about 129
Find Beats And Mark command 198
Find Beats commands 90
5.1 surround sound 213
flange, defined 280
Flanger effect 144
Flush Virtual File command 58
flushing
  defined 281
  forcing complete flush 48
Force New Row command 15
frames, snapping to 91
frequencies
  selecting 88
  viewing 85
Frequency Band Splitter effect 157
frequency range, analyzing 119
frequency, defined 281
Full command 170
Full Reverb effect 150
Function tab, in Favorites dialog box 254
FX Mixer window 186

G

Generate commands 106
graph controls 29
Graphic Equalizer effect 131
Graphic Phase Shifter effect 145

Downloaded from www.Manualslib.com manuals search engine

ug.book Page 295 Tuesday, March 16, 2004 1:29 PM

grid lines, showing or hiding 50
Group Waveform Normalize command 244

H

Hard Limiting effect 137
Help 3
Help tab, Favorites dialog box 255
Hertz (Hz), defined 281
hidden clips, revealing 179
Hiss Reduction effect 127
horizontal scroll bar 19
Hybrid tool 164, 168

I

IFF format 233
importing
  audio 61
  files 25
  video files 208
IMPS folder 154
impulse
  and the Convolution effect 154
  and the Full Reverb effect 151
  defined 281
Impulse (.imp) files 154
Index cue
  about 98
  defined 281
input devices
  adjusting record levels 82
  choosing 36
  monitoring levels 80
  setting properties 38
input gain
  adjusting 82
  monitoring 80
Insert Audio From Video command 208
Insert In CD Project command 258
Insert Video command 208
Insert/Delete Time command 163
inserting
  empty audio clips 178
  files into tracks 25
interleaved 6-channel Windows Media 9 Pro (WMA) 223
interpolate, defined 281
inverting audio 106
ISRC number 260

K

keyboard shortcuts 12

L

latency, correcting 39
left channel
  editing 92
  selecting data in 88
Level Meters
  defined 282
  using 80
levels, adjusting 82
LFE 217
limiter, defined 282
limiting 246
Linear Energy Plot 50
linking sessions to Adobe Premiere Pro and After Effects 51
Load Meter 165
Logarithmic Energy Plot 50
Loop Duplicate command 173
Loop Info tab 200
Loop Properties command 202
loops
  about 197
  defined 282
  defining 198
  extending or shortening 205
  key of 200, 202, 204
  previewing 77
  properties in Edit View 200
  properties in Multitrack View 202
  stretching to fit 201
  synchronizing 205
  tempo of 200, 202, 204
lossless encoding 239

M

macros 243
magnifying analysis graphs 122
magnifying the display 17
Manage Temporary Folder Reserve Space command 57
Marquee Selection tool 88
mastering
  audio 117
  CDs 257
  defined 282
memory, allocating buffers 46, 54
menus 12
Merge This Take (Destructive) command 75
Merge/Rejoin Split command 172
metadata 116
metronome 204
Microsoft ACM 233
Microsoft DirectX 214

Downloaded from www.Manualslib.com manuals search engine

ug.book Page 296 Tuesday, March 16, 2004 1:29 PM

MIDI
defined 282
fundamentals of 273
tracks, about 185
triggering 12
MIDI devices, choosing 36
MIDI Timecode (MTC), defined 282
MIDI Trigger, defined 282
millisecond (ms), defined 282
miniplug, defined 283
Mix Down commands 196
Mix Down To CD Project command 258
Mix Gauge 165
defined 283
Mix Paste command 94, 113
mixdowns
defined 283
exporting to audio 229
exporting to video 230
Mixers window 190
mixing
automating with clip envelopes 188
default bit-resolution 56
defined 283
preferences 54
m-Law Wave format 232
Monitor Record Level command 80
mono waveforms, converting to stereo 112
mono, defined 283
mouse
changing cursor 45
zooming with mouse wheel 45
Move/Copy Clip tool 164, 168
moving windows 14
mp3Pro format 237
Multichannel Encoder
dialog box 213
elements of 214
Pan Envelope automation 218
panning sound 215
Preview Device menu 222
Preview Format menu 223
previewing 221
Multichannel Encoder command 215
multichannel WMA 225
Multitap Delay effect 149
Multitrack View
preferences 54
switching to 11
using 10
work area of 162
multitrack-only effects 156
Music effect 155
muting
audio 103
tracks 181

N

navigating in display window 17
new files, creating 84
New Session command 162
Next/Sun format 238
noise
generating 108
noise gate, defined 283
Noise Reduction effect 128
noise shaping
curve, for dithering 38
defined 283
when dithering 114
nondestructive editing, defined 283
Normalize effect 137
Normalize tab, Group Waveform Normalize dialog box 246
normalizing
defined 284
files for CDs 257
groups of files 244
Notch filter effect 131
Nyquist Frequency, defined 284

O

offline editing, defined 284
online Help 3
Open Append command 62
Open As command 62
Open Audio From Video command 208
Open command 61
Open Session command 63
opening files 61
order, defined 284
Organizer window 24
out-of-band peaks, working with 246
output devices
adjusting playback levels 82
choosing 36
setting properties 37
output gain
adjusting 82
monitoring 80
overdrive 155

Downloaded from www.Manualslib.com manuals search engine

ug.book Page 297 Tuesday, March 16, 2004 1:29 PM

P

p.d.f. (probability distribution function) 38
pan envelopes 188, 218
Pan/Expand effect 140
Panner Point, in Multichannel Encoder 218
panning
  in sessions 56
  in surround 215
Parametric Equalizer effect 132
pasting
  audio 93
  resampling data 53
pauses, adding between CD tracks 260
pausing playing or recording 79
PCM Raw Data format 241
PCM, defined 284
peak amplitude, displayed in Level Meters 80
peak files
  defined 284
  specifying options for 51
performance, measuring with Mix Gauge and Load Meter 165
phase
  analyzing 121
  defined 284
pink noise
  defined 285
  generating 108
Pitch Bender effect 145
pitch, changing 145
placekeeper windows 16
play lists 102
Play Looped button 199
playback cursor 69
playback devices. *See output devices*
playing audio
  about 75
  automatically 62, 77
  choosing a device for 36
  downsampling 38
  setting the start point 68
plug-in
  defined 285
  effects, using 32
Power Indicator lines 217
pre-emphasis, for CDs 261
preferences 43
premade loops 197
preroll and postroll, for effect previews 31
presets
  defined 285
  using for effects 28
previewing
  audio 61
  audio in Organizer window 77
  defined 285
previewing effects
  buffer size for 46
  in Edit View 31
process signal, for Vocoder effect 158
punch-in
  defined 285
  deleting unused takes 56
punching in audio 73

Q

quantization, defined 285
Quick Filter effect 133
QuickVerb effect 151

R

RAM, allocating buffers 46, 54
ranges
  saving to new file 227
  selecting in Multitrack View 164
RAW format 232, 241
RCA cable, defined 286
real-time effects
  about 185
  applying 185
  locking tracks with 188
  mixing 186
  removing 185
real-time, defined 286
recording audio
  32-bit options 39
  about 72
  choosing a device for 36
  correcting for drift 55
  setting the start point 68
recording devices. *See input devices*
redoing edits 22
referenced clip, defined 286
reflective surfaces, simulating with Echo effect 148
Refresh Effects List command 33
Refresh Now command 165
Remove All Tracks command 259
Remove Clips command 179
Remove Selected Tracks command 259
removing noise 125
renaming files 249
repositioning windows 14
resample, defined 286
Rescale Volume Envelopes command 188
reserve space 57

Downloaded from www.Manualslib.com manuals search engine

ug.book Page 298 Tuesday, March 16, 2004 1:29 PM

resizing windows 14
restoring audio, about 117
restoring default workspace 44
reverb
  adding 150
  defined 286
  tails 152
Reverb effect 152
reversing audio 106
Revert To Saved command 23
rewinding 79
ReWire tracks
  mixing down 196
ReWire tracks, working with 184
right channel, editing 92
right-click menus 12
rip, defined 286
ripping, from CD 64

S

S/N ratio, defined 287
SAM format 232
sample rate
  changing 111
  considerations for choosing 110
  defined 287
  previewing a different rate 111
sample type
  converting 61, 110
  displaying in status bar 21
sample, defined 287
sampler, defined 287
SampleVision format 238
sampling rate, fundamentals of 270
saving
  audio files 227
  mixdowns to audio 229
  mixdowns to video 230
  sessions 228
Scientific Filters effect 133
.scp files 248
script collection (.scp) files 248
Script tab, in Favorites dialog box 255
scripts
  about 243
  working with 249
Scripts command 250
scrolling
  about 19
  preferences in Edit View 44
  preferences in Multitrack View 57
Select All Tracks command 259
Select Entire Wave command 88
selecting audio 87
Selection/View Controls 20
separator bars 254
sequencer, defined 287
Session Properties command 204
sessions
  about 161
  converting the sample rate 229
  creating 162
  default session 163
  defined 287
  deleting time in 163
  inserting clips 63
  inserting time in 163
  linking to Adobe Premiere Pro or After Effects 207
  opening 63
  recording into 73
  saving 228
  using as SMPTE masters or slaves 166
Set Current Clipboard command 92
setting up Adobe Audition 35
Settings command 43
shortcuts 12
Show View Tabs command 11
showing and hiding
  toolbars 13
  Transport Controls 71
  view tabs 11
  windows 15
shrinking audio 146, 177
silence
  deleting 104
  generating 103
64-bit doubles (RAW) format 232
slip editing clips 170
smoothing edit boundaries 52
SMP format 238
SMPTE
  defined 287
  monitoring synchronization 22
  using sessions as masters or slaves 166
SMPTE Master Enable command 166
SMPTE Slave Enable command 166
SMPTE Start Offset command 166
Snap To Clips command 205
Snap to Frames command 209
Snap To Loop Endpoints command 205

Downloaded from www.Manualslib.com manuals search engine

ug.book Page 299 Tuesday, March 16, 2004 1:29 PM

Snap To Ruler (Coarse) command 205
Snap To Zero Crossings command 198
snapping
  about 91
  clips 170
SND format 233, 238
soloing tracks 181
sound card
  defined 287
  positioning information 46
sound wave, defined 287
sound, fundamentals of 267
special effects
  convolution 153
  distortion 153
  music 153
  noise 108
  tones 109
spectral display, adjusting 49
Spectral View
  adjusting display settings 50
  selecting frequencies in 88
  switching between Waveform View and 85
  using 85
splines 29
Split command 172
SPTI (SCSI Pass Through Interface) 46
Static Peaks option 81
Statistics command 123
Status Bar window 21
stereo
  defined 287
  imagery, changing 139
Stereo Field Rotate effect 141
stereo waveforms
  choosing which channel to edit 92
  converting to mono 112
stop playing or recording 78
Stretch effect 146
stretching audio 146, 177
stripe, defined 287
Studio Reverb effect 153
subwoofer 217
Surround Panner 216, 217
SVX format 233
Sweeping Phaser effect 144
symmetric dithering 52
Synchronize Clips With Edit View command 69
Synchronize Cursor Across Windows command 69
synchronizing
  loops 205
  multitrack latency 39

T

Take History command 75
tempo
  defined 288
  editing 71
temporary folders
  managing size of 57
  setting location 47
time display format 70
Time Selection tool 164, 168
time stretching clips 177
Time window 69
timecode, defined 288
Timed Record Mode command 72
timeline, snapping to ruler 91
tones, generating 109
Tool tab, Favorites dialog box 255
toolbars 13
tooltips 13
touch tones, generating 106
track controls, defined 288
track cues
  defined 288
  for CD 98
  inserting 258
Track Equalizers window 182
Track Mixer 191
Track Properties window 180
tracks
  bit depth, setting 182
  changing volume 180
  channel, setting 182
  defined 288
  equalizing 182
  inserting files into 25
  locking 188
  muting 181
  naming 180
  panning 180
  soloing 181
tracks for CDs
  editing source audio 260
  inserting 258
  removing 259
  selecting 259
  setting properties 260
Transport Controls
  Fast Forward button 79
  Go to Beginning button 79
  Go to End button 79
  Pause button 79
  Play buttons 75
  Record button 72, 73, 74

Downloaded from www.Manualslib.com manuals search engine

ug.book Page 300 Tuesday, March 16, 2004 1:29 PM

Rewind button 79
showing and hiding 71
Stop button 78
Trim command 95, 170
Trim Digital Silence command 102
TXT format 234

U

undocking windows 14
undoing edits
about 22
preferences for 47
unity gain, defined 288
UPC/EAN, defined 262
upsampling quality level 53

V

Variable Bit Rate (VBR) 225
VBR 225
VBR encoding 237, 239
vertical ruler
changing scale of 86
scrolling in 19
video
clips 209
exporting mixdowns to 230
preparing mixdowns for export 211
previews, customizing 210
working with 207
video files, importing audio and video from 208
view tabs, showing and hiding 11
views, switching between 11
VOC format 235
Vocoder effect 158
volume envelopes 188
volume,making consistent 244
VOX format 235
VST plug-ins 32

W

WAV format 232, 233, 236, 239
Wave Cache 46
wave file, defined 288
Wave Properties command 116, 200
waveform clip, defined 288
waveform display
defined 289
view modes for 85
waveform statistics, viewing 123
Waveform View
adjusting display settings 50
switching between Spectral View and 85
waveform, defined 288
waveforms
considerations for editing 83
fundamentals of 268
viewing 85
wet, defined 289
white noise
defined 289
generating 108
window handle 14
Windows Media 9 Pro 213
Windows Notepad 250, 252
Windows PCM format 225, 239
Windows Recording Mixer command 82
windows, using 14
WMA format 239
workspace
changing colors 48
changing display settings 50
described 9
restoring to default 44
writing CDs 261

Z

zero crossing, defined 289
zero-crossing points
adjusting selections to 89
automatically adjusting edits to 57
snapping to 91
Zoom commands in Multichannel Encoder 221
zooming 17
zooming analysis graphs 122

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